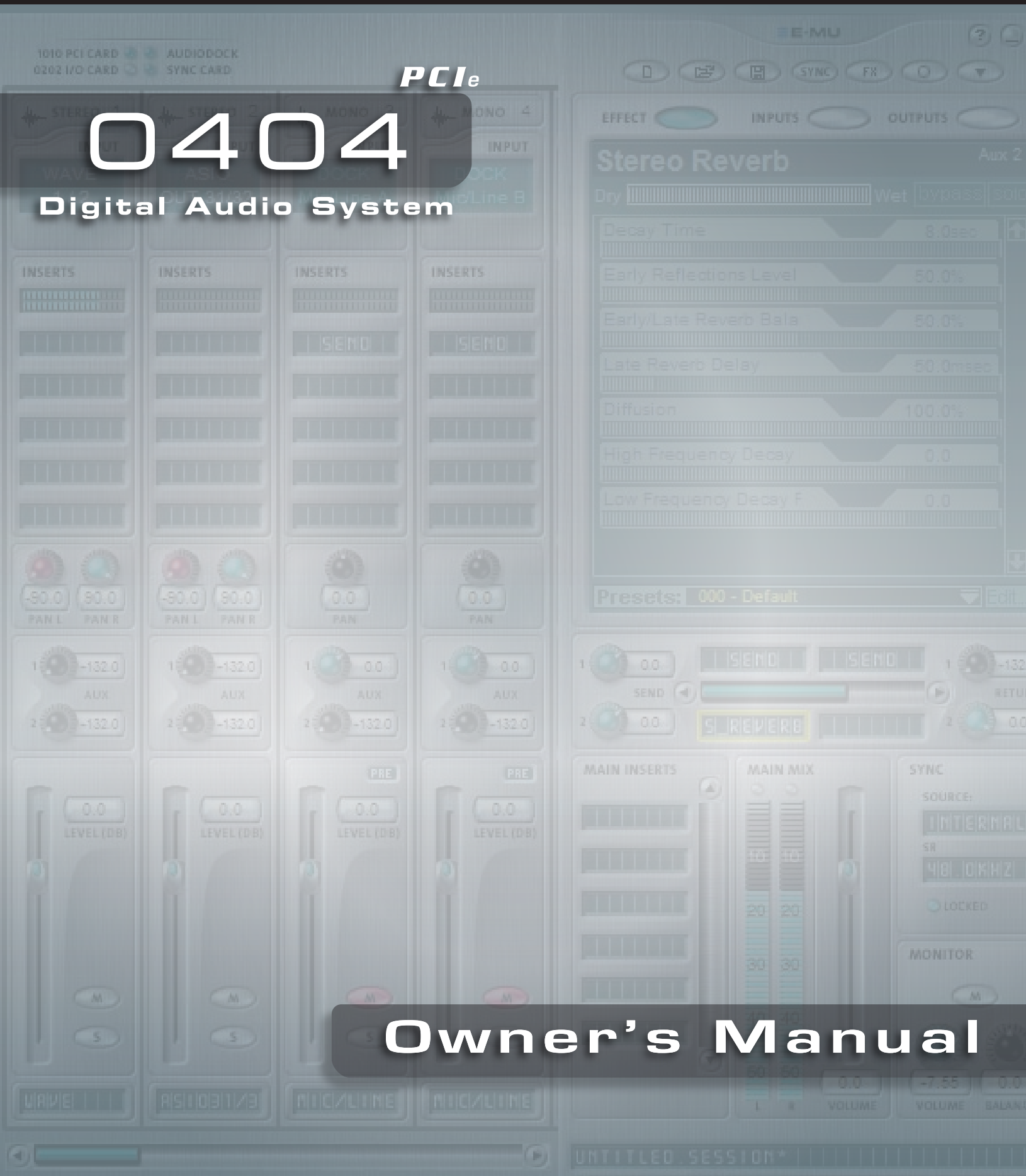
PCIe

0404

Digital Audio System



Owner's Manual

E-MU

E-MU 0404 PCIe
Digital Audio System

Owner's Manual

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Software Version: 2.10

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1- Introduction

Welcome!

Thank you for purchasing the E-MU 0404 digital audio system. Your computer is about to be transformed into a professional quality audio processing workstation. We've designed your E-MU Digital Audio System to be logical, intuitive and above all, to provide you with pristine sound quality. This system offers unprecedented value by providing studio-quality, 24-bit/192kHz multi-channel recording and playback at an astounding price.

E-MU Digital Audio System Components

E-MU 0404	Inputs & Outputs
▪ E-MU 0404 PCIe Card ▪ Analog Breakout Cable (1/4"(1) Analog Breakout Cable (RCA)) ▪ RCA Jack Breakout Cable ▪ Digital Breakout Cable ▪ E-MU Digital Audio System Software/Driver Install CD-ROM ▪ Production Tools Software Bundle CD-ROM ▪ Quick Start Guide	(2) Ch. S/PDIF Optical In (2) Ch. S/PDIF Optical Out or (2) Ch. S/PDIF Coaxial In (2) Ch. S/PDIF Coaxial Out (1) MIDI Input & Output (16 ch.) (2) 24-bit unbalanced Line Inputs (2) 24-bit unbalanced Line Outputs

The System Includes:

The E-MU 0404 PCIe Card provides 2 line level, unbalanced analog inputs, 2 line level, unbalanced analog outputs, plus MIDI input and output. This is a finely-tuned audio interface, using high performance 24-bit/192kHz A/D - D/A converters to deliver an unbelievable 111dB of dynamic range. Check out the complete specs on page 99.

The PCIe card contains a powerful hardware DSP processor which allows you to use over 16 simultaneous hardware-based effects, which place minimal load on your computer's CPU. The E-MU 0404 PCIe Card also provides a S/PDIF stereo digital input and output with either optical or coaxial connections. A built-in MIDI interface allows you to connect external MIDI instruments or keyboards directly to your computer.

The PatchMix DSP mixer application is included in all the systems. PatchMix DSP delivers unmatched flexibility in routing your audio between physical inputs and outputs, virtual (ASIO) inputs and outputs and internal hardware effects and buses—no external mixer needed. You can add digital effects, EQs, meters, level controls and ASIO sends anywhere you like in the signal chain.

Because the effects and mixing are hardware-based, there is no latency when you record. You can even record a dry signal while monitoring yourself with effects! Mixer setups can be saved and instantly recalled for specific purposes such as recording, mixdown, special effect setups or general computer use.

You'll want to keep up with the latest software and options for your E-MU digital audio system. You can find all of this, plus other helpful information, at the E-MU Website: http://www.emu.com.

Notes, Tips and Warnings

Items of special interest are presented in this document as notes, tips and warnings.

- **Notes** provide additional information related to the topic being discussed. Often, notes describe the interaction between the topic and some other aspect of the system.
- **Tips** describe applications for the topic under discussion.
- **Warnings** are especially important, since they help you avoid activities that can cause damage to your files, your computer or yourself.

Legacy Sync Daughter Card

The legacy Sync Daughter Card is NOT compatible with the 0404PCIe card. The Sync Daughter Card was an option for the 0404 PCI card that provided Word Clock in and out, SMPTE in and out, and MIDI Time Code output.

2 - Installation

Setting Up the Digital Audio System

There are five basic steps to installing your E-MU system:

1. IMPORTANT - Remove any other sound cards you have in your computer. (Once you are sure that the E-MU card works properly, your old sound card can be reinstalled if desired. Bear in mind that depending on your computer system, multiple sound cards do not always work together.)
2. Install the 0404 PCIe card in your computer. Go there.
3. Attach the Analog and Digital breakout cables to the rear of the 0404 card.
4. Install the PatchMix DSP software onto your computer.
5. Connect audio and MIDI cables between the 0404 PCIe and your other gear.

Notes for Installation

- IF AT ANY TIME DURING THIS INSTALLATION YOU SEE NO RESPONSE: Use the Alt-Tab feature to select other applications. One of them may be the Microsoft Digital Signature warning. It is possible for this warning to appear behind the installation screen.
- Make sure you have the latest Windows Service Packs from Microsoft (Windows XP - SP 2 or higher, Vista - SP 1 or higher).
- Disable onboard sound and uninstall all other sound cards. (If you wish to try using multiple sound cards in your system, do so after you have confirmed that your E-MU Digital Audio System is operating normally.)
- InstallShield "IKernel Application Error" on Windows XP: When installing this software on Windows XP, you may be confronted with a "kernel error" at the very end of installation. This is an issue with the InstallShield program, which is what we use to install software on your computer. Please do not be alarmed by this, as the error is innocuous.

  To read more about this error, and obtain instructions on how to avoid getting the message, please visit this website: http://support.installshield.com/kb/view.asp?articleid=q108020
- Multiple Digital Audio System sound cards are currently not supported.

Please read the following sections as they apply to your system as you install the E-MU 0404, paying special attention to the various warnings they include.

Safety First!

▼ *As you install hardware components, observe the following general precautions to avoid damage to your equipment and yourself.*

- *To avoid possible permanent damage to your hardware, make sure that all connections are made to the E-MU 0404 PCIe card with the host computer's power off.* ***Unplug the computer's power cable to make sure that the computer is not in sleep mode***.
- *Take care to avoid static damage to any components of your system. Internal computer surfaces, the E-MU 0404 PCIe board and the interfaces are susceptible to electrostatic discharge, commonly known as "static." Electrostatic discharge can damage or destroy electronic devices. Follow these procedures when handling electronic devices in order to minimize the possibility of causing electrostatic damage:*
- *Avoid any unnecessary movement, such as scuffing your feet when handling electronic devices, since most movement can generate additional charges of static electricity.*
- *Minimize the handling of the PCIe card. Keep it in its static-free package until needed. Transport or store the board only in its protective package.*
- *When handling a PCIe card, avoid touching its connector pins. Try to handle the board by its edges only.*
- *Before installing a PCIe card into your computer, you should be grounded. Use a ground strap to discharge any static electric charge built up on your body. The ground strap attaches to your wrist and any unpainted metal surface within your computer. If you don't have a ground strap, you can ground yourself by touching the metal case of another piece of grounded equipment.*
- *Before connecting a cable to your interface or between PCIe cards, touch the connector sleeve of the cable to the sleeve of the jack to which you'll be connecting the cable in order to discharge any static build-up.*

Connector Types

These connector types are used to connect the E-MU 0404 hardware components. They will be referred to by the name shown in the first column of the following chart:

Name	Description	Connects
DB-15 Digital	Digital Cable Connector	0404 PCIe card and Digital I/O
DB-9 Analog	Analog Cable Connector	0404 PCIe card and Analog I/O
1/4" Jacks	1/4" Breakout Cable	2 channel analog input/output
RCA Jacks	RCA Breakout Cable	2 channel analog input/output
S/PDIF In	RCA Connector	S/PDIF digital audio devices
S/PDIF Out	RCA Connector	S/PDIF digital audio devices
S/PDIF Optical In	TOSLINK Optical Connector	S/PDIF digital audio devices
S/PDIF Optical Out	TOSLINK Optical Connector	S/PDIF digital audio devices

Installing the E-MU 0404 PCIe Card

Note: This installation is very simple, but if you are not familiar with the installation of computer peripherals and add-in boards, please contact your authorized E-MU Systems dealer or an approved computer service center to arrange for the installation.

▶ To install the 0404 PCIe card into your computer

1. Make sure that the power switch on your computer is off.
   IMPORTANT: Unplug the power cord from the wall outlet!
2. Touch a metal plate on your computer to ground yourself and to discharge any static electricity.
3. Follow the computer manufacturer's recommended procedure for opening the case.
4. Remove the metal bracket from one PCIe x1 slot as shown in figure 1 below. *(PCIe x1 slots are the smallest of the PCIe slots.)* Put any screws you removed (if applicable) aside for later.

■ **Note:** Some computer cases don't use screws to secure PCIe cards. In this case, follow the instructions that came with your computer.

Figure 1

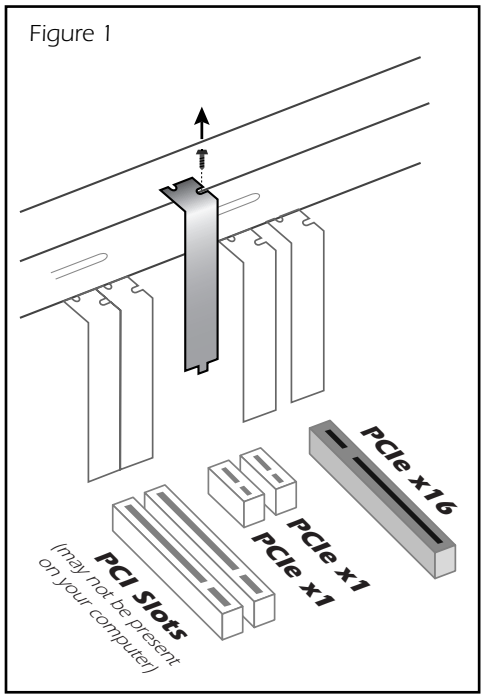
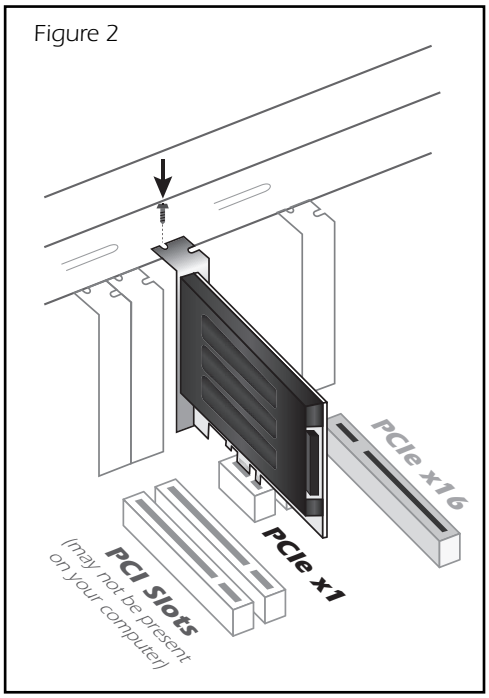


Figure 2



5. Align the E-MU 0404 PCIe card with the slot and press gently but firmly down into the slot as shown in figure 2.
6. Do not force the E-MU 0404 card into the slot. Make sure that the gold finger connector of the card is aligned with the PCIe bus connector on the motherboard before you insert the card into the PCIe slot. If it doesn't fit properly, gently remove it and try again.
7. Secure the card into the slot using one of the screws (if applicable) you placed aside earlier.
8. Attach the Analog and Digital breakout cables to the rear of the 0404 card.
9. Connect your audio cables to the breakout cables.

Software Installation

Installing the E-MU 0404 Drivers

The first time you restart your PC after installing the E-MU 0404 PCIe card, you will need to install the PatchMix DSP software and E-MU 0404 PCIe card drivers.

Windows XP, Windows XP x64, Windows Vista, Windows Vista x64

The software is not compatible with other versions of Windows.

1. After you have installed your audio card, turn on your computer. Windows automatically detects your audio card and searches for device drivers.
2. IMPORTANT: When prompted by Windows for the audio drivers, click the Cancel button.
3. Insert the E-MU software Installation CD into your CD-ROM drive. If Windows AutoPlay mode is enabled for your CD-ROM drive, the CD starts running automatically. If not, from your Windows desktop, click Start->Run and type d:\setup.exe (replace d:\ with the drive letter of your CD-ROM drive). You can also open the CD and double-click Setup.exe.
4. The installation splash screen appears. Follow the instructions on the screen to complete the installation.
5. Choose "Continue Anyway" when you encounter the "Windows Logo Testing" warning screen. See note below for more information.
6. When prompted, restart your computer.

■ ***Serial Number -*** *During the registration process, you will be asked to enter your serial number. The serial number is located on the back of the box and on the 0404 PCIe Card.*

Note About Windows Logo Testing

When you install the Digital Audio System drivers, you will see a dialog box informing you either that the driver has not been certified by Windows Hardware Quality Labs (WHQL), or that the driver is signed by Creative Labs, Inc, and you will be asked if you would like to continue with the installation.

The Digital Audio System audio drivers are not certified by WHQL because the product does not support some of the features that the Microsoft Windows Logo Program requires, most notably Universal Audio Architecture (UAA) and Digital Rights Management (DRM).

Despite this, the Digital Audio System audio drivers have been rigorously tested using the same test procedures that a WHQL qualified driver requires, and it passes in all of the other important categories, including those that measure the relative stability of the driver. So, it is perfectly safe to install these drivers on your computer.

Uninstalling all Audio Drivers and Applications

At times you may need to uninstall or reinstall some or all of the audio card's applications and device drivers to correct problems, change configurations, or upgrade outdated drivers or applications. Before you begin, close all audio card applications. Applications still running during the uninstallation will not be removed.

1. Click Start -> Settings -> Control Panel.
2. Double-click the Add/Remove Programs icon.
3. Click the Install/Uninstall tab (or Change or Remove Programs button).
4. Select the E-MU 0404 PCIe card entry, or the application entry and then click the Add/Remove (or Change/Remove) button.
5. In the InstallShield Wizard dialog box, select the Remove option.
6. Click the Yes button. Restart your computer when prompted.
7. You may now re-install existing or updated E-MU 0404 PCIe card device drivers or applications.

3 - PCIe Card & Interfaces

The E-MU 0404 PCIe Card

The E-MU 0404 PCIe card contains E-MU's powerful E-DSP chip which leaves more power free on your CPU for additional software plug-ins and other tasks. Bit depth is controlled by your recording or audio application. The 0404 PCIe card always sends and receives 24-bit audio.

DB-9 & DB-15 Connectors

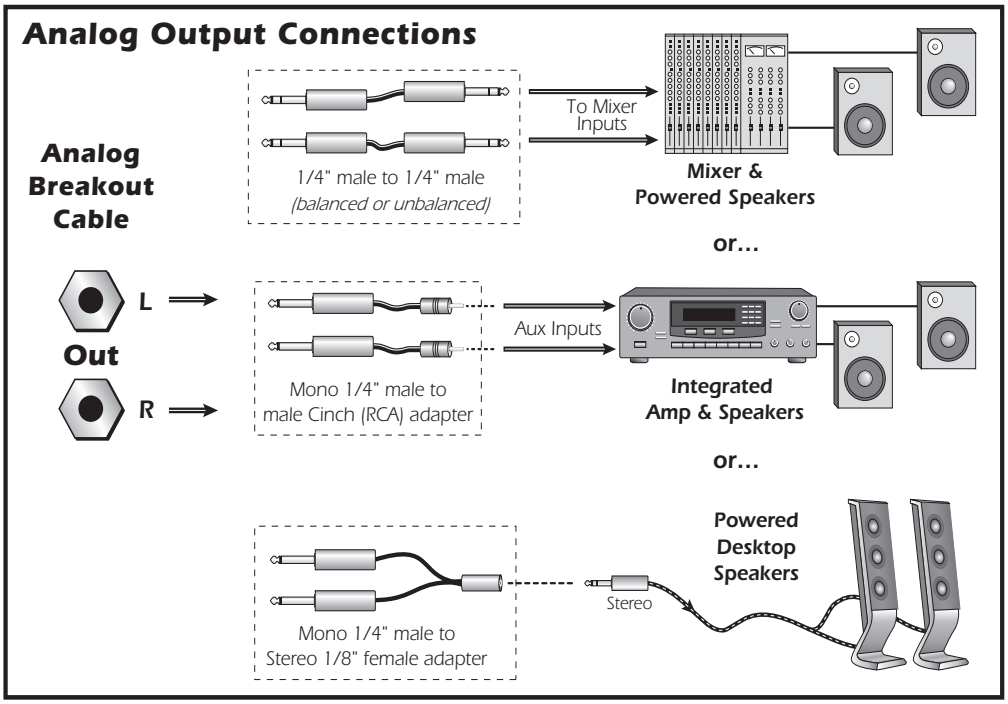
Connects the analog and digital breakout cables to the 0404 PCIe card. Connect the 1/4" Analog Breakout Cable to the DB-9 connector and the Digital Breakout Cable to the DB-15 connector.

Analog Breakout Cable

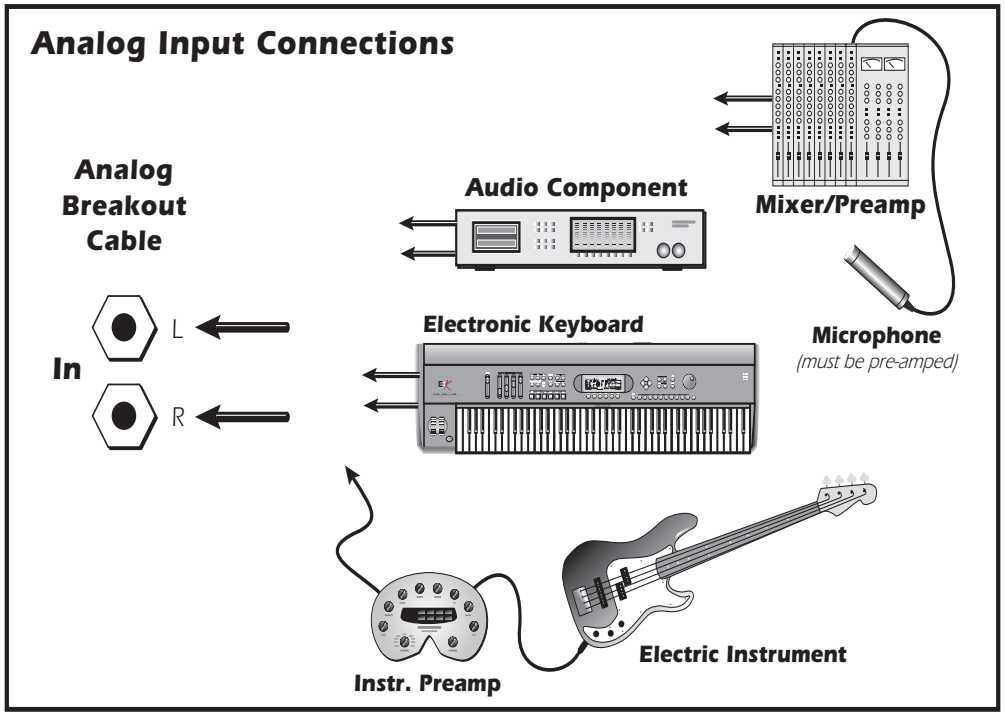
The 0404 PCIe card provides one pair of 24-bit unbalanced analog inputs and one pair of 24-bit unbalanced analog outputs. The analog breakout cable is designed to accommodate 1/4" plugsRCA (phono) plugs. Use 1/4" to RCA adapter cables to connect to consumer audio gear.

Note: You may have to combine adapters to connect desktop computer speakers. An adapter with two male 1/4" phone plugs to 1/8" stereo female adapter is available directly from E-MU Systems in the US, or from your local electronics shop in other countries.

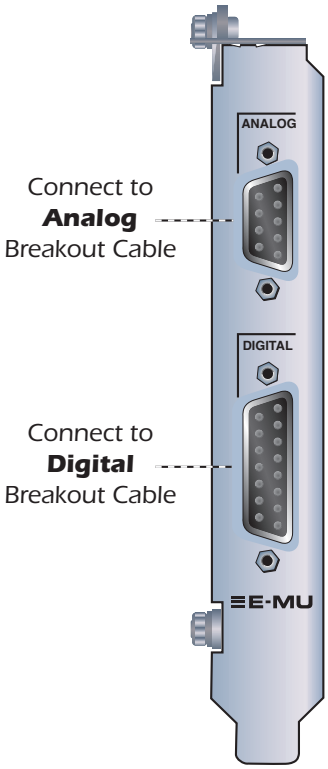
The outputs can feed any line level input such as a mixing board, the auxiliary input on your stereo, or a set of powered speakers. They are not designed to drive headphones directly. Use a mixer, home stereo receiver, or headphone amplifier to monitor with headphones.



The inputs can be connected to any line level stereo signal from keyboards, CD-players, cassette decks, etc. Use 1/4" to RCA adapter cables to connect to consumer audio gear.



Digital Breakout Cable



S/PDIF Digital Audio Input & Output

RCA phono jacks are the standard coaxial connectors used for S/PDIF (Sony/Philips Digital InterFace) connections. A single jack carries two channels of digital audio. The E-MU 0404 receives digital audio data with word lengths of up to 24-bits. Data is always transmitted at 24-bits.

S/PDIF digital I/O allows you to receive and/ or transmit of digital data from external digital devices such as a DAT, external analog-to-digital converters or external signal processors equipped with digital inputs and outputs.

S/PDIF can also be transmitted and received via the TOSLINK optical connectors on the Digital Breakout Cable. Optical connections have certain advantages such as immunity to electrical interference and ground loops. Make sure to use high quality glass fiber light pipes for connections longer than 1.5 meters.

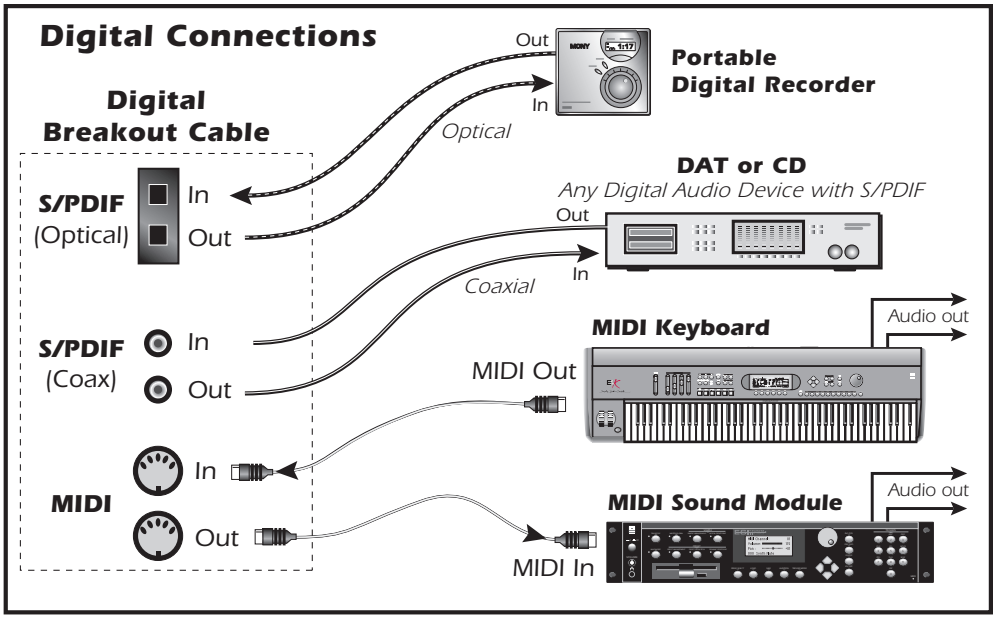
The optical S/PDIF and RCA coaxial S/PDIF Inputs cannot be used simultaneously, however BOTH S/PDIF Outputs are available simultaneously (carrying the same signal). See System Settings.

The S/PDIF out can be configured as either Professional or Consumer mode in the **Session Settings** menu. The

0404 PCIe card can be connected to AES/EBU digital audio systems through the use of a cable adapter. See AES/EBU to S/PDIF Cable Adapter for details.

The S/PDIF input and outputs are usable at the 44.1kHz, 48kHz and 96kHz sample rates. The word clock contained in the input data stream can be used as a word clock source. See Using External Clock.

▼ **Important:** When using any type of digital I/O such as S/PDIF, you MUST sample sync the two devices or clicks and pops in the audio See Using External Clock

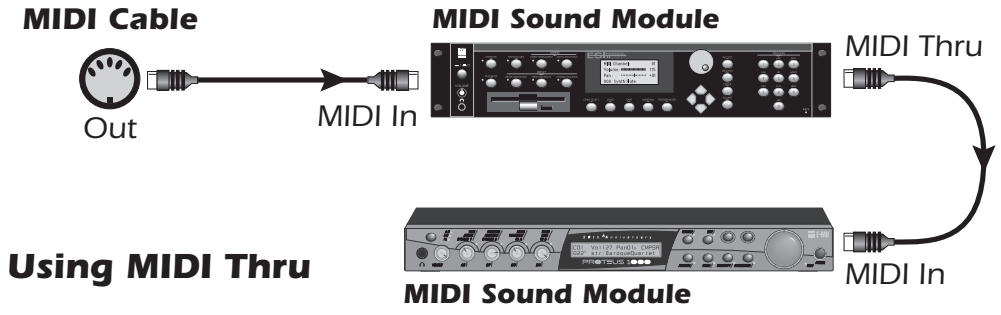


MIDI (**M**usical **I**nstrument **D**igital **I**nterface) is a standard specification for networking two or more devices together. Connect **MIDI Out** to external MIDI instruments and connect **MIDI In** to a controller such as a MIDI keyboard.

Unlike S/PDIF, the MIDI cable does not carry audio data. In its most basic application, MIDI tells a synthesizer when to start and stop playing specific notes. MIDI also carries other information such as how hard the note was played, what sound to play, the channel volume and many other commands. The most important thing to remember is that MIDI contains CONTROL information, not the sound itself.

Information on the MIDI cable can be assigned to any one of sixteen channels so that different musical lines can be assigned to play specific sounds or MIDI instruments.

To connect more than one MIDI instrument to the 0404 PCIe card, the MIDI Thru port on your synthesizer can be used. MIDI Thru carries an exact copy of the data on the synthesizer's MIDI input port.



4 - The PatchMix DSP Mixer

PatchMix DSP

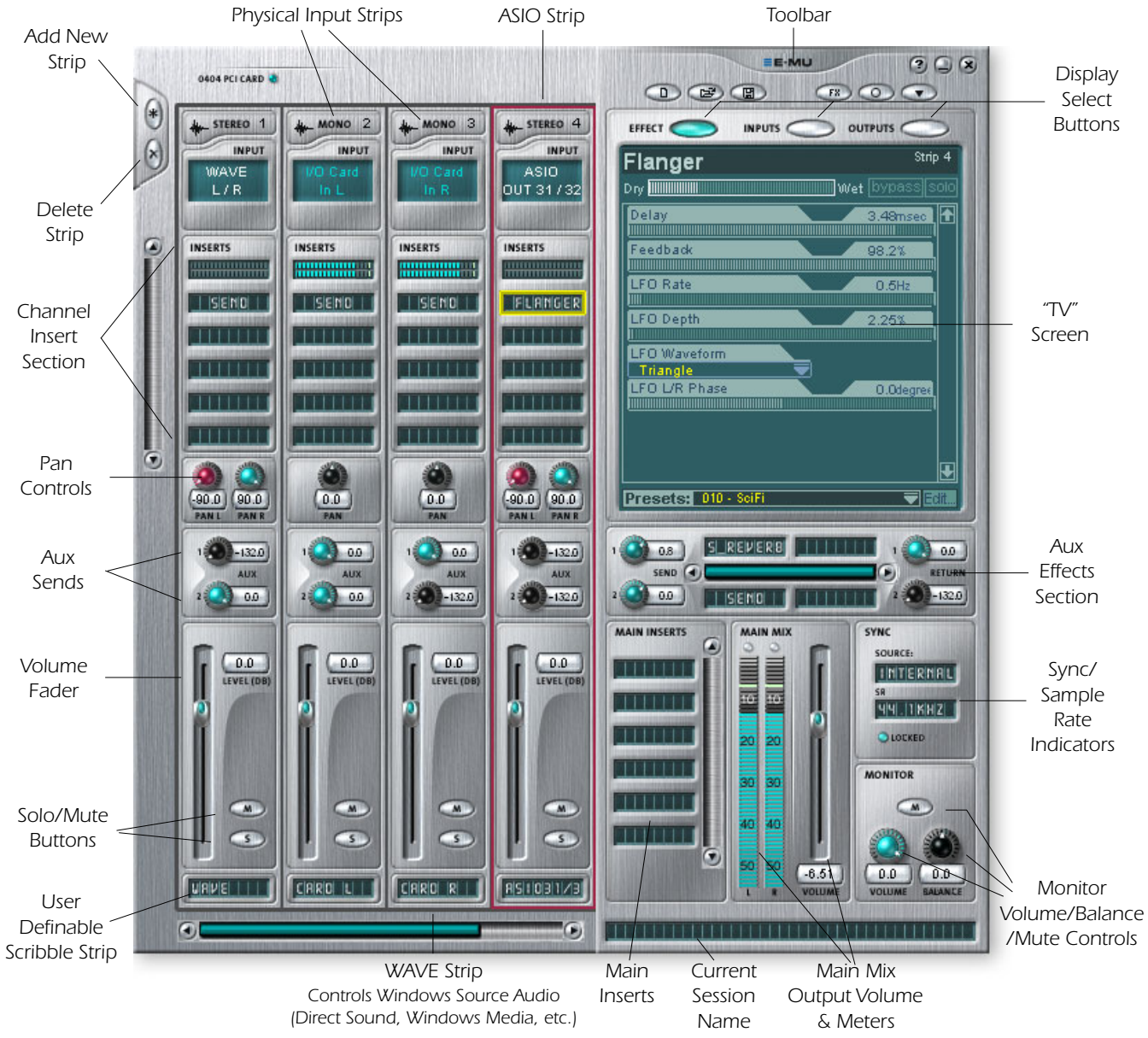
The PatchMix DSP Mixer is a virtual console, which performs all of the functions of a typical hardware mixer and then adds a few new tricks of its own. PatchMix DSP greatly simplifies audio operations such as ASIO/WAVE routing, volume control, stereo panning, equalization, effect processing, effect send/return routing, main mix and monitor control, without getting in the way of your other software. It's easy and it works…beautifully!

▶ To Invoke the PatchMix DSP Mixer

1. Left-click once on the E-MU icon on the Windows System Tray. The PatchMix DSP mixer window appears.

● Click on the buttons and knobs in the mixer screen below to jump to the description of the control.

Overview of the Mixer



Mixer Window

The Mixer consists of four main sections.

Application Toolbar — Lets you manage sessions and show/hide the various views.

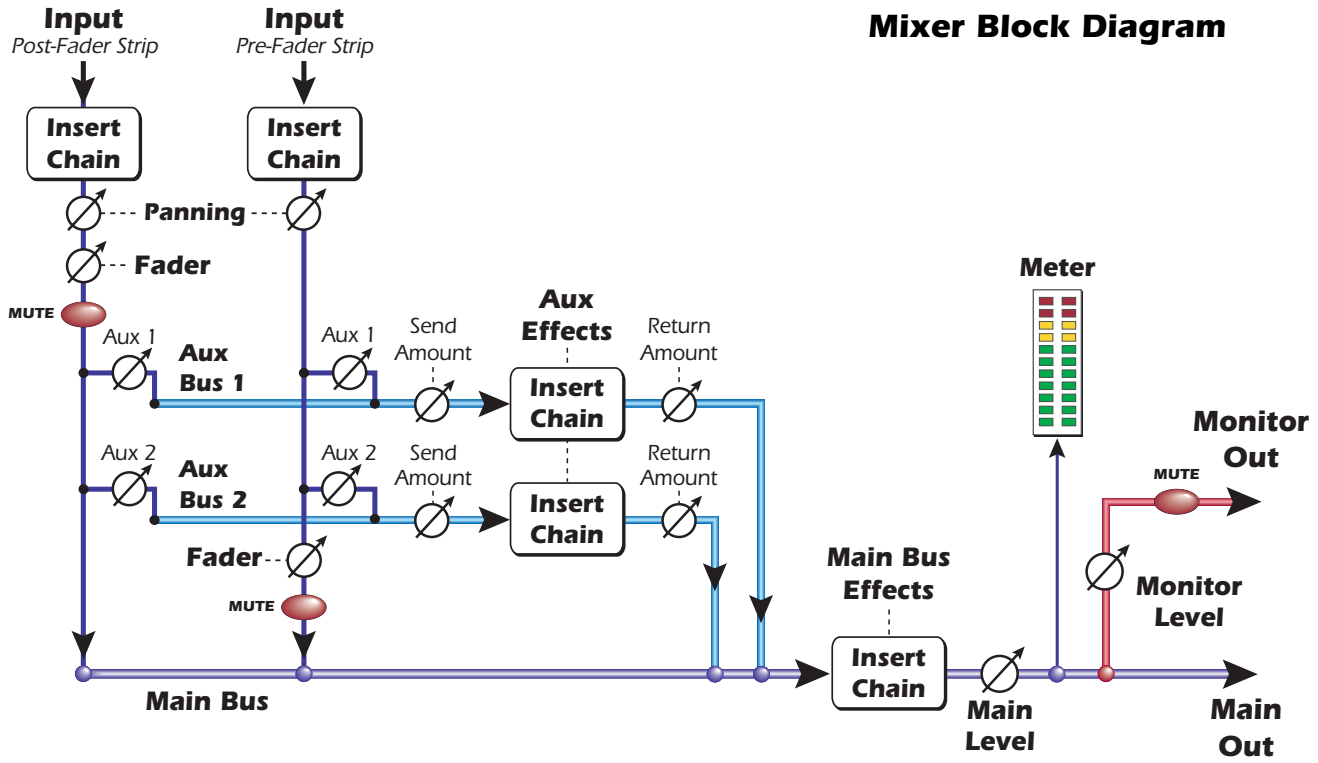
Main Section — Controls all the main levels, aux buses, and their inserts. This section also has a "TV" which shows parameters for the currently selected effect and the input/output patchbay. It also shows the session's current sample rate and whether the Digital Audio System is set to internal or external clock.

Mixer Strips — This section is located to the left of the Main Section and shows all the currently instantiated mixer strips. Mixer strips can represent **Physical** analog/digital inputs, or **Host** inputs such as ASIO or Direct Sound. Mixer strips can be added or deleted as necessary. This section can be resized by dragging the left edge of the frame.

Effects Palette — This popup window is invoked by pressing the FX button in the toolbar. Effects presets are shown here, organized by category. From this window, you can drag and drop effect presets into the insert slots available on the mixer strips and main section aux buses and main inserts.

● **Tip:** Study the diagram below to understand how the PatchMix DSP Mixer

A simplified diagram of the mixer is shown below.

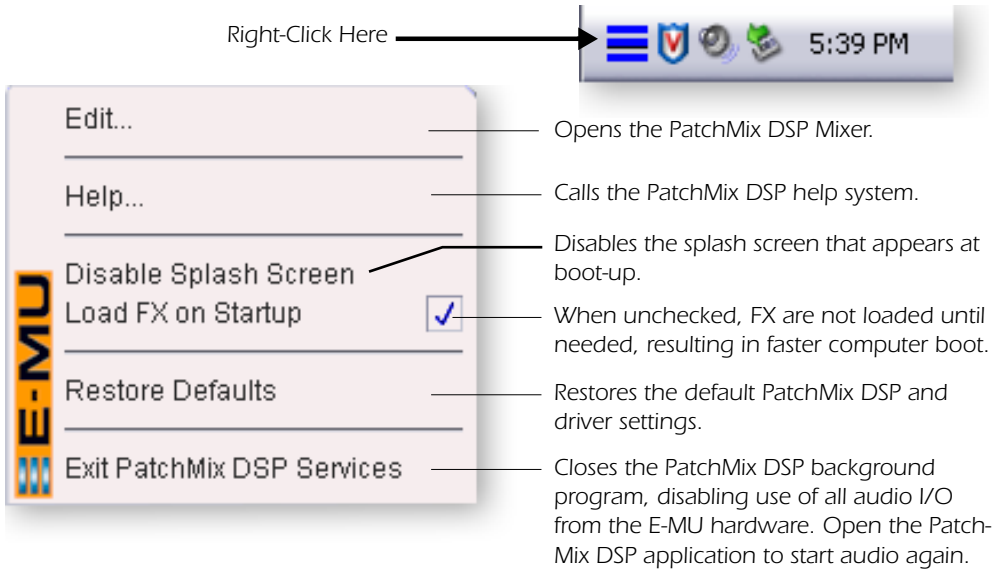


Pre Fader or Post Fader

When creating a new Mixer Strip, you have the option for the Aux Sends to be Post Fader (both Aux Sends come after the channel fader) or Pre Fader (both Aux Sends come before the channel fader). The Pre-fader option allows you to use either Aux Send as another mix bus, which is unaffected by the channel fader. More Information.

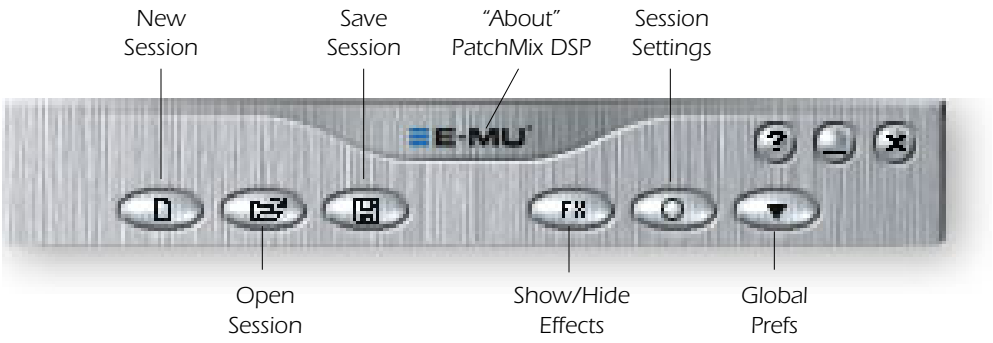
E-MU Icon in the Windows Taskbar

Right-clicking on the E-MU icon in the Windows taskbar calls up the following window.


Right-Click Here
5:39 PM
Edit...
Help...
Disable Splash Screen
Load FX on Startup
Restore Defaults
Exit PatchMix DSP Services
E-MU
Opens the PatchMix DSP Mixer.
Calls the PatchMix DSP help system.
Disables the splash screen that appears at boot-up.
When unchecked, FX are not loaded until needed, resulting in faster computer boot.
Restores the default PatchMix DSP and driver settings.
Closes the PatchMix DSP background program, disabling use of all audio I/O from the E-MU hardware. Open the Patch-Mix DSP application to start audio again.

● **Restore Defaults**: Always try this option first if PatchMix is crashing or if you are having any other strange audio problems.

The Toolbar



● Click the buttons in the toolbar to learn about their function.

New Session	Calls up the "New Session" dialog box. New Session.
Open Session	Calls up the standard "Open" dialog box, allowing you to open a saved Session.
Save Session	Calls up the standard "Save" or "Save As…" dialog boxes, allowing you to save the current Session.
Show/Hide Effects	Toggle button that shows or hides the FX palette.
Session Settings	Calls up the Sessions Settings window. Session Settings.
Global Preferences	Calls up the Global Preferences window.
About PatchMix DSP	Right-Click on the E-MU logo to view the "About PatchMix DSP" screen, which provides the software and firmware version numbers and other information.

The Session

The current state of the PatchMix DSP mixer (fader settings, effects routings…everything!) can be saved as a Session. Whenever you create or modify a mixer setup, all you have to do is Save it to be able to recall it at a later time.

Before you begin using PatchMix DSP, you need to set it up to be compatible with the other software applications you may be running. The most important consideration is your system sample rate. **PatchMix DSP and any applications or other digital gear you are using must be set to the same sample rate.** PatchMix DSP can run at 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4 kHz or 192kHz sample rates, but the effect processors are only available at the 44.1kHz or 48kHz rates.

When you start a new PatchMix DSP Session, the first choice you make is to select the sample rate. Once set, you can only easily switch between 44.1kHz and 48kHz. You cannot switch between 44/48kHz and 88k/96k/176k/192k. **With a change to these higher sample rates, you must start a new session.**

You can also set up an external sync source, thereby obtaining the sample rate from some other device or application. External sync can be obtained from the S/PDIF input. If the session is set at 44.1kHz or 48kHz and the external source is coming in at 96kHz, the Sync Indicator will be extinguished (off), but PatchMix will attempt to receive the external data. if the Sync Indicator is Off, the two units are NOT sample locked and you should correct this condition to avoid intermittent clicks in the audio. **Always check for the presence of the "LOCKED" indicator whenever you are using a digital interface.**

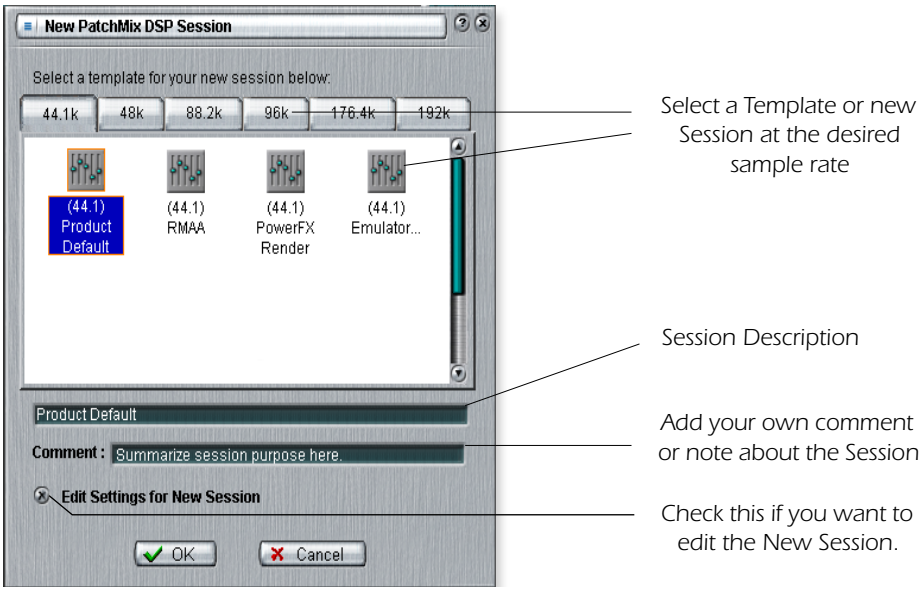
▼ **Important:** *When using any form of digital input, you MUST synchronize the Digital Audio System to the external digital device.* See Using External Clock

PatchMix DSP comes with several session templates to choose from so when you create a new session you can either create a "blank" session based around a designated sample rate, or select from a list of template starting points.

In a PatchMix DSP session the number of strips in the mixer is dynamically configurable. See Pre Fader or Post Fader. This allows you to create only those strips you need up to a maximum number determined by available DSP resources and available inputs.

New Session

You create a new session by clicking the "New Session" button in the PatchMix DSP main Toolbar. The following dialog box appears.



You can now select one of the factory template sessions. The factory templates are pre-programmed with specific setups such as audio recording or mixing. The selector tabs categorize Template Sessions into three groups based on sample rate, 44.1k/48k, 88k/96k or 176k/192k.

You can create your own templates by simply copying or saving sessions into the "Session Templates" folder (Program Files\Creative Professional\Digital Audio System\E-MU PatchMix DSP\Session Templates).

The "Session Path" allows you to choose the destination for your Session. The default location is in the "My Sessions" folder within the "My Documents" folder.

There is also a Comment area that you can use to give yourself some clue as to what you were thinking when you created the session.

Open Session

To Open a saved session, click on the Open Session button. A dialog box appears allowing you to choose one of your saved Sessions to open. Choose one of your saved sessions and click on the Open button.

Save Session

To Save a session, click on the Save Session button. A Save dialog box appears allowing you to choose a location in which to save the current Session. The "My Sessions" folder is chosen by default.

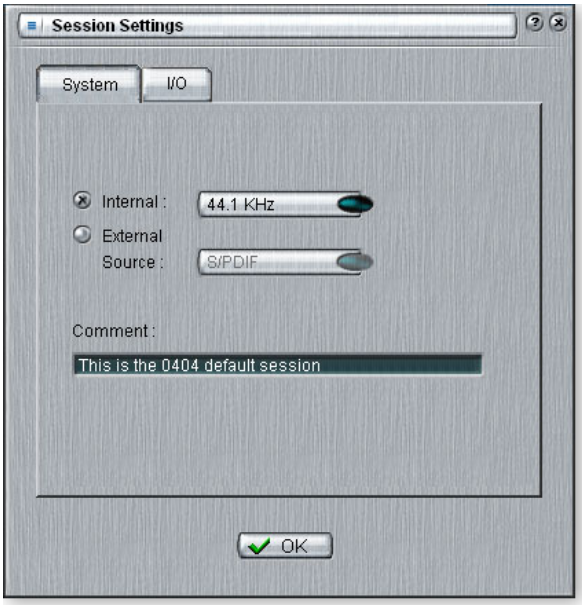
Get in the habit of saving the session whenever you have created a special mixer setup. This will make your life much easier as you can recall a setup for many different audio modes such as: recording, mixing, special ASIO routings, etc.

● *Saving a session "defragments" the effect/DSP resources. If you have used all your effects and need another, try saving the session.*

Session Settings

System Settings

Pressing the Session Settings button on the toolbar brings up the System Settings window shown below. Click the tabs to select System or I/O options.



The System Settings include the following choices:

▪ **Internal/External Clock**	Selects between **internal** or **external** clock source as the master clock source for the system
▪ **Sample Rate**	Selects the sample rate when using internal clock. Your choices are: **44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz, 192kHz**
▪ **External Clock Source** (ext. clock only)	Select ***S/PDIF*** as an external sample clock source.

■ **Note:** if set to "External" without an external clock present, PatchMix DSP defaults to the internal 48kHz clock rate.

Using External Clock

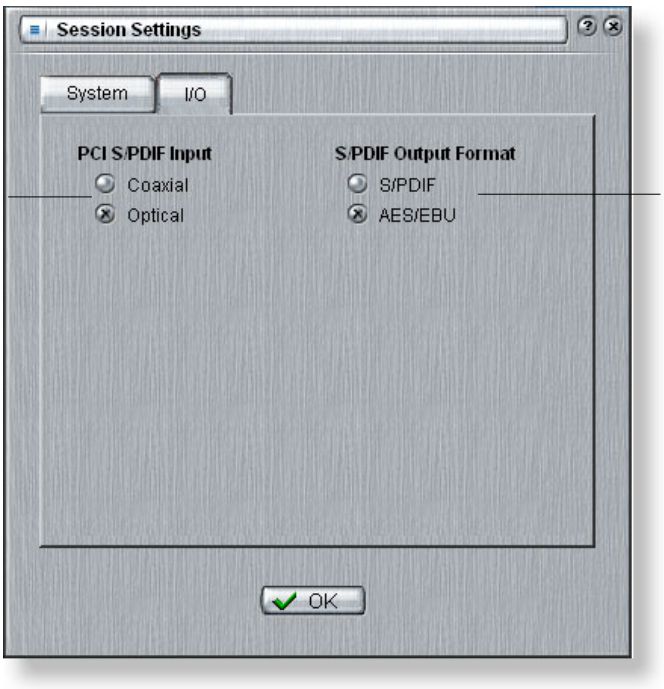
Whenever you are connecting two or more devices using digital I/O such as S/PDIF, one of the digital devices MUST supply the master clock to the others. This master clock runs at the system sample rate and can be distributed using a dedicated cable (word clock) or embedded into a data stream such as S/PDIF. Common symptoms of unsynced digital audio include, random clicks or pops in the audio or failure of the digital stream to be recognized. Always check for the presence of the "LOCKED" indicator whenever you are using a digital interface.

If an External Clock is interrupted or switched after the Session has been created (except between 44.1k <-> 48k), the "LOCKED" indicator will extinguish and PatchMix will attempt to receive the external data. The two units are NOT sample locked however, and you should correct this condition to avoid intermittent clicks in the audio.

I/O Settings

The 0404 PCIe card is optimized for signal levels of -10dBV (consumer standard) for the analog inputs and outputs. -10dBV levels are compatible with most consumer audio gear.

Setting correct input and output levels is important! You can measure the level of an input by inserting a meter into the first effect location in the strip. Adjust your external equipment outputs for the optimum signal level. See "To Set the Input Levels of a Strip" for details.



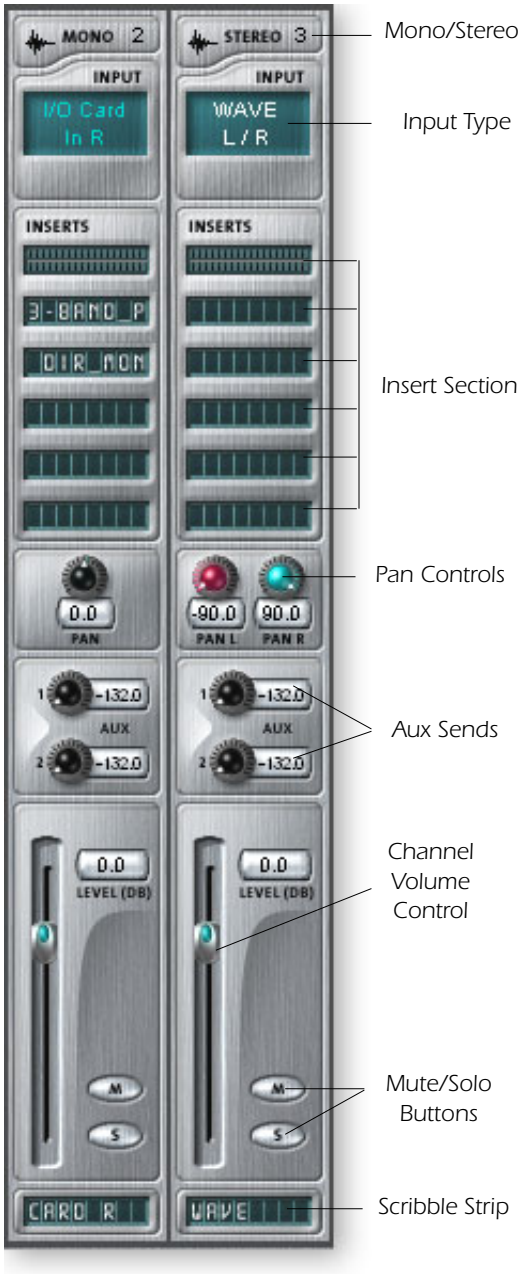
The optical digital TOSLINK input and output on the Digital Breakout Cable can be used to transmit and receive stereo S/PDIF.

▪ **PCIe Card S/PDIF Input**	Selects between coaxial or optical S/PDIF input. S/PDIF out is always transmitted on both the coaxial and optical outputs.
▪ **S/PDIF Output Format**	Selects between S/PDIF or AES/EBU format for S/PDIF. This sets the S/PDIF-AES status bit, but does not affect the signal level.

Input Mixer Strips

PatchMix DSP Line Input Mixer Strips are mono. The WAVE and S/PDIF strips are stereo. Each input mixer strip can be divided into four basic sections.

▪ Insert Section	Effects, EQ, External/Host Sends and Returns can be inserted into the signal path.
▪ Pan Controls	This control positions the signal in the stereo sound field.
▪ Aux Sends	Used to send the signal to sidechain effects or create separate mixes.
▪ Volume Control	Controls the output level of the channel.



This screen shows a mono strip on the left and a stereo strip on the right.

Input Type

The very top of the strip is labeled mono or stereo and displays the type of the assigned input. Input mixer strips can be added as desired and can be configured to input the following:

- **Physical input** *(Analog/SPDIF)*
- **Host Input** *(Direct Sound, WAV, ASIO source)*

● The **Input Type** will turn RED if the input is not available.

● Physical input strips are shown with BLUE text.

● **Host** input strips are shown with WHITE text.

Inserts

You can drag and drop effects from the Effects Palette or Right-click to insert a Physical or ASIO Send or Send/Return A Peak Meter, Trim Control or Test Signal can also be inserted by Right-clicking.

Pan Controls

These controls allow to you position the channel in the stereo sound field. Dual controls on stereo strips allow you to position each side independently.

Aux Sends

These controls send the signal to sidechain effect processors such as reverb and delay. They can also be used to create separate mixes for the artist or for recording.

Volume Control

Controls the output level of the strip into the main/monitor mix bus.

Mute/Solo Buttons

These convenient buttons allow you to solo or mute selected channels.

Scribble Strips

Click inside the scribble strip and type a name of up to eight characters.

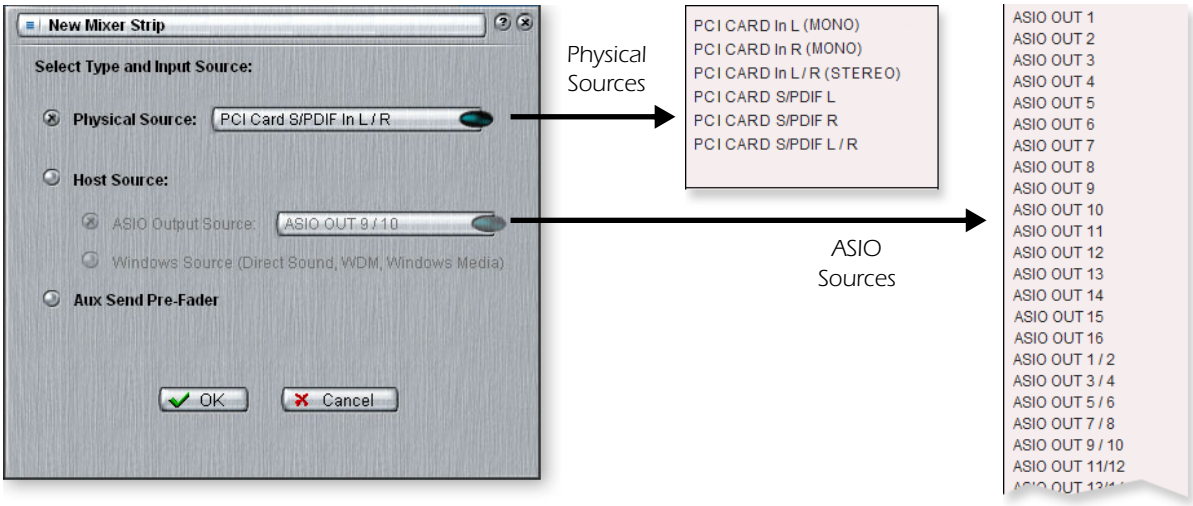
Mixer Strip Creation

PatchMix DSP is a dynamically configurable mixer. Each mixer session can contain an arbitrary number of channel strips up to a limit set by the number of available input sources and available DSP resources.

You must create a strip for each mono or stereo audio input, and for each ASIO stream you wish to use in your software application. This is important because outputs will not appear in your software application until you have created ASIO strips in PatchMix.

▶ To Add a New Strip:

1. Click on the New Mixer Strip button. See Overview of the Mixer
2. The Assign Mixer Strip Input Dialog appears:



3. Select the desired input to the mixer strip from the following choices:

- **Physical Source:** Stereo analog or digital card input (Analog or S/PDIF)
- **Host - ASIO Source input** Streaming audio from an ASIO software application.
- **Host - WAVE input** Windows sound sources—WAVE, Direct Sound, WDM, CD

Mixer Strip Type	Function
Physical: PCIe Card Analog	24-bit mono or stereo analog inputs.
Physical: PCIe Card S/PDIF	2 channel digital audio from the S/PDIF input.
HOST SOURCE	**Function**
Host ASIO Output Source From software application	Mono or stereo digital audio from an ASIO source (i.e. a recording or other software app). ASIO Out 1-32, ASIO Out 1/2. 3/4, etc.
Host Windows Source From Windows	Direct Sound, WDM, Windows Media *(Sound generated or handled by Windows.)* WAVE 1/2 - Default stereo source such as game sound, CD player, beep sounds, etc. WAVE 3/4 - Additional WDM channels

4. Select Pre-Fader Aux Sends or leave the box unchecked for Post-Fader Aux Sends.
5. Click **OK** to create a new strip or **Cancel** to cancel the operation.

● **CDs & MP3s:** The **WAVE 1/2** strip is used to playback CDs, Windows Media Player, and Direct Sound.

■ 2 channels of WAVE recording and 4 channels of multichannel WAVE playback are supported.

- Classic MME
- DirectSound
- Direct WDM / Kernel Streaming (KS)

▼ 192kHz/96kHz DVD-Audio disks are protected against digital copying. Most DVD-Audio disks contain duplicate 48kHz audio tracks which will play back on the 0404.

▶ To Delete a Mixer Strip:

1. Click the top of the mixer strip you wish to delete. A red border appears around the strip, indicating that it is selected.
2. Click on the Delete Mixer Strip button or right-click and choose Delete, or use the Delete key on the PC keyboard. See Overview of the Mixer

Multichannel WAVE Files

The 0404 PCIe supports **2 channels of WAVE recording** and **4 channels of multichannel WAVE playback**. The WAVE channels are available for the following types of WDM devices:

- Classic MME
- DirectSound
- Direct WDM / Kernel Streaming (KS)

DirectSound and the **WDM/KS** interfaces allow up to four channels of Wave Out while the classic MME interface only exposes 2 channels.

The WAVE channels operate at all sample rates. For additional information about WDM behavior at high sample rates, see page 49.

192kHz/96kHz DVD-Audio disks are protected against digital copying. Most DVD-Audio disks contain duplicate 48kHz audio tracks which will play back on the 0404.

Select **DirectSound** as the output format when using Windows Media Player and other DVD player applications.

Insert Section

The **Insert Section** is next in line. PatchMix DSP effects can be selected from the Effects Palette and dropped into the insert locations. See "The Effects Palette". Any number of effects can be inserted in series. **The signal flow through the insert section is fromTop to Bottom.** If a DSP effect is *above* a Send, the effect will be applied to that Send. If the DSP effect is placed *below* the Send, the send will be dry (no effects).

The Inserts also have the unique ability to patch into ASIO/WAVE and external equipment. Special inserts, ASIO/WAVE Sends, External Sends and External Send/Returns can be dropped into the insert section to route the signal anywhere you want.

The Insert/Patch Bay is incredibly flexible. Want to send the strip to your audio recorder? Just insert an ASIO send into the insert section and select the ASIO pair you want. That's it! That input is available in your ASIO software.

● *You have to create an ASIO strip or ASIO Send in order to activate these ASIO channels in your software.*

Suppose you wanted to record a submix of your voice mixed with a CD? Simply place a HOST ASIO SEND into the Aux Insert section and turn up the Aux Sends on the input channels you want in the mix. (This patch is actually shown in the Mixer Overview on page 18.) Note that the WAVE L/R Strip and the I/O Card In L strip are routed to Aux Send 2, which has a HOST ASIO SEND insert to the recording application.

The following types of inserts can be selected.

Hardware Effect	Reverb, EQ, Compressor, Flanger, etc. using PatchMix DSP's effects which do not load your CPU.
ASIO Send	Splits off the signal and sends to an ASIO host input such as a software audio recorder or anything that uses ASIO.
ASIO Direct Monitor	Sends the signal to a selected ASIO host input, then returns a selected ASIO host output to the chain. Use for recording with "ASIO Direct Monitor Send/Return"
Ext. Send/Return	Sends the signal to a selected external output, then returns it to the chain via a physical input.
External Send	Sends the signal to an external output. See "To Add a Send Insert:".
Peak Meter	Peak meters allow you to monitor the signal level anywhere in the chain. See "Meter Inserts".
Trim Pot	You can insert a gain control with up to 30 dB of gain or attenuation. A peak level meter and phase inverter are also included. See page 33.
Test Tone	This special insert outputs a calibrated sine wave or noise source, which can be used to track down audio problems. See page 33.

Working with Inserts

The Inserts are one of most powerful features of the PatchMix DSP system as they allow you to configure the mixer for a wide variety of applications.

▼ **Problem:** *You are not able to select analog return when using the Send/Return Inserts.*

Reason: *The 0404 Digital Audio System only supports* ***Stereo I/O*** *in the Send/Return Insert section. The analog inputs on the 0404 are mono.*

▶ To Add an Effect to an Insert Location:

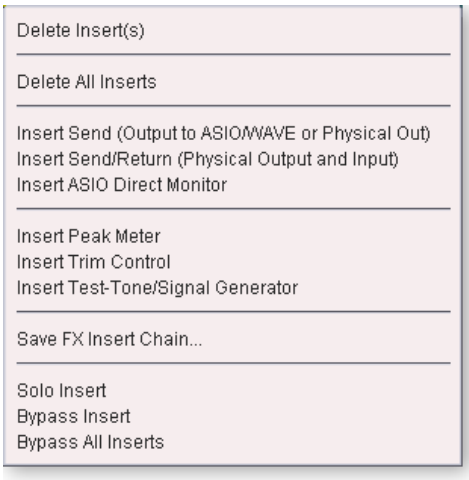
1. Press the FX button. The effects palette appears.
2. The effects are organized into categories. Click on a folder to open it.
3. Select the effect you want, drag it over the insert section, then drop it into an insert location.
4. To rearrange the order of effects, simply drag and drop them into the desired order.

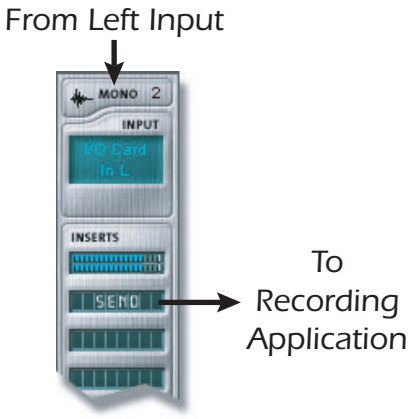
Note: The Physical Output & Input option is "grayed-out" when using the default Session

Reason: The 0404 Digital Audio System has only 4 physical inputs and 4 physical outputs. The Send/Return option is grayed-out because all the physical I/O resources available for send/return have been used in this Session. If S/PDIF I/O is not being used elsewhere, it becomes available in the Send/Return list.

The Insert Menu

Right-Clicking over the insert section brings up a pop-up selection box containing various insert options to help you control and manage your inserts.





To connect an input to your recording software: ***Add a Host ASIO Insert.***

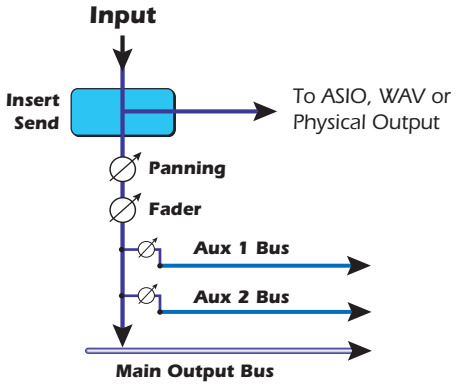
■ **Note**: The order of the inserts is important since the signal flows from Top to Bottom.

For example, if an ASIO Insert is placed *above* an reverb effect, the ASIO signal will be sent to your recording app *without* reverb. If the reverb was placed above the ASIO Send, the ASIO signal to your recording app will have reverb applied.

▶ To Add a Send Insert:

This type of insert send *splits* the signal at the insert point and sends it out to the selected destination. (An "ASIO Send" becomes an input on your recording application, a "Physical Out" goes to a pair of output jacks. the signal also continues down the strip to the Aux Sends and main mixer outputs.)

1. Right-Click over the Insert section. A pop-up dialog box appears.
2. Select "Insert Send (to ASIO/WAVE or other)" from the list of options. The following dialog box appears.

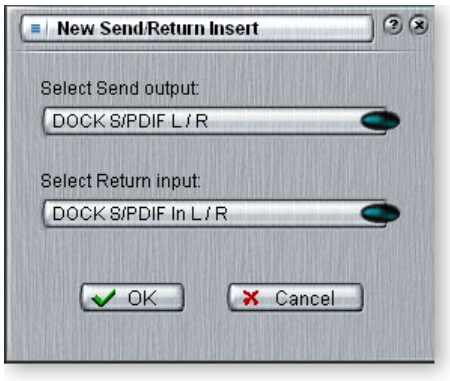


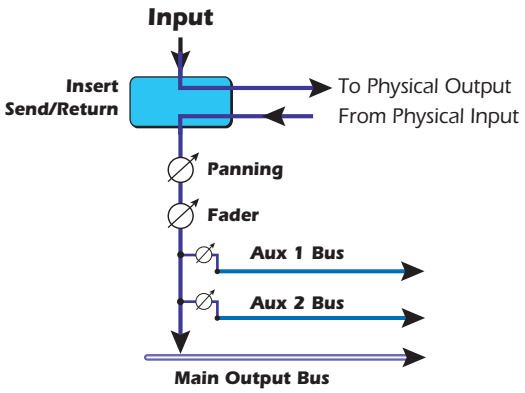
3. Choose one of the Send Outputs. Click on a destination to select it.
4. Click OK to select the output or Cancel to cancel the operation.

▶ To Add a Send/Return Insert:

This type of insert send *breaks* the signal at the insert point and sends it out to the selected destination such as an external effect processor. A return source signal is also selected which *returns* the signal to the channel strip after processing.

1. Right-Click over the Insert section. A pop-up dialog box appears.
2. Select "Insert Send/Return" from the list of options. The following dialog box appears.





▼ If the source or destination you want to use is not available in the list, they are probably already being used elsewhere. Check the input Strips, Inserts and Output Assignments.

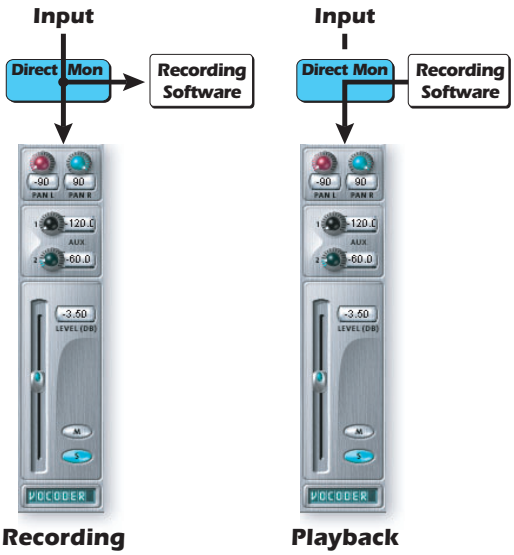
3. Choose one of the Send Outputs. Click on a destination to select it.
4. Choose one of the Return Inputs. Click on a source to select it.
5. Click OK to select the Send and Return or Cancel to cancel the operation.

ASIO Direct Monitor Send/Return

This type of insert send *breaks* the signal at the insert point and sends it out to the selected ASIO Host Input destination (such as Cubasis). A return source signal is also selected which *returns* the signal to the channel strip from an ASIO Host Output.

The ASIO Direct Monitor Send/Return is unique in that it utilizes ASIO 2.0 zero-latency monitoring. **In order to utilize this feature, Direct Monitoring must be enabled in the audio recording application.**

While recording, the Direct Monitor Send/Return routes the signal to the recording application, but monitors directly from the input to eliminate latency. During playback, the recording application automatically switches the Direct Monitor Send/Return to monitor the recorded track.



The Direct Monitor Send/Return also allows the recording application to control volume and pan. Normally when using direct monitor recording you'll want to control the volume and pan from the recording application. In this case, set the PatchMix DSP stereo pan controls hard left and right, mono pan controls to center, and the fader to 0dB.

▶ To Add an ASIO Direct Monitor Send/Return:

1. Right-Click over the Insert section. A pop-up dialog box appears.
2. Select "Insert ASIO Direct Monitor" from the option list. The following dialog box appears.

3. Choose one of the Send Outputs. Click on a destination to select it.
4. Choose one of the Return Inputs. Click on a source to select it.
5. Click OK to select the Send and Return or Cancel to cancel the operation.

Meter Inserts

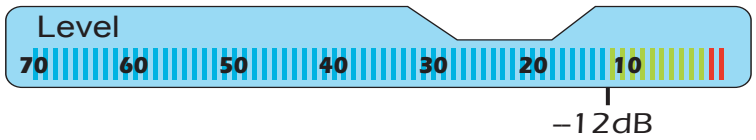
Keeping track of signal levels is important in any audio system, be it analog or digital. You want to keep the signal levels running as close to maximum in order to achieve high resolution and low noise. On the other hand, you don't want the signal level so high as to cause clipping. To help you maintain optimum signal levels, we have included Peak Level Meters, which can be dropped into any insert location.

The insert meters are of the "peak hold" type. The topmost bar in the meter holds its highest level for a second to let you see transients that would otherwise be too quick for the eye. A numeric readout above the meter shows the peak-hold level in dB.

The peak meters are also color-coded to indicate the signal strength. The chart below outlines the meanings of the colors. Avoid lighting the topmost red bar, as this indicates distortion of the signal. Click on the clip indicator to turn it off.

Meter Color	Indicates
Red	Indicates signal clipping.
Yellow	Good strong signal level.
Green	Signal is present.

One of the most obvious uses of the insert meters is to set input levels. On the analog inputs, the analog-to-digital converter (ADC) is one of the most critical points in the signal path. You want the input signal level to drive the 24-bit ADCs into their optimum range without clipping. A reading of 0dB on an input meter indicates signal clipping.



Each bar of the meter equals 1dB. The yellow bars begin at -12dB below full scale.

The insert meters are also useful to monitor incoming digital signals such as ASIO or S/PDIF to make sure the mixer is receiving a proper signal level. They're also great for trouble-shooting, since you can place them virtually anywhere in the mixer.

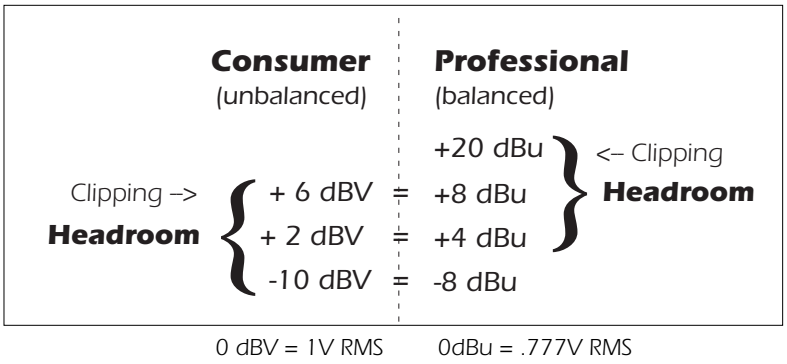
▶ To Insert a Meter

1. Right-Click on an Insert location of the mixer strip. A pop-up dialog box appears.
2. Select "Insert Peak Meter." A stereo peak meter appears in the insert location.
3. Select FX in the Main Section, then Left-Click on the meter insert. The meters are now shown in high resolution in the TV screen.

To Set the Input Levels of a Strip

1. Select the topmost Insert location on a mixer strip and insert a meter (see above).
2. Left-click on the meter insert to see the meter in the TV screen.
3. Feed your audio signal to the input of the mixer strip. The meter should now show the signal level.
4. Adjust the output level of the external device (synthesizer, instrument, preamp, etc.) feeding the 0404 Card. The meter should be in the yellow region most of the time with occasional forays into the red. If the clip indicator ever comes on, reduce the signal level.

Comparison of -10dBV & +4dBu Signal Levels



Making the Best Possible Recording

Making a good digital recording is easier than ever thanks to the high resolution 24-bit A-D converters on your Digital Audio System. These converters are much more forgiving than the 12-bit or 16-bit converters of the past. Even so, to get the best performance possible, you'll need to follow a few basic guidelines.

First, whenever you input an analog signal to the Digital Audio System, make sure that you're feeding the A-D converters with an optimum signal level. The quality of a digital recording is directly related to the signal level you feed into the A-D converters. If the analog input level is set too low, you lose resolution—if it's set too high, the A-D converters will clip.

To measure the input level, simply add an insert meter to the channel strip in PatchMix DSP. These meters are accurately calibrated to display 1dB for each bar on the meter. You can enlarge the meter view by clicking on the insert meter in a strip and selecting the “Effect” button at the top of the TV screen.

In order to supply the correct input level, you’ll need to adjust the output of your analog source (electric instrument or preamp) so that the input level comes close to 0dB without ever going over.

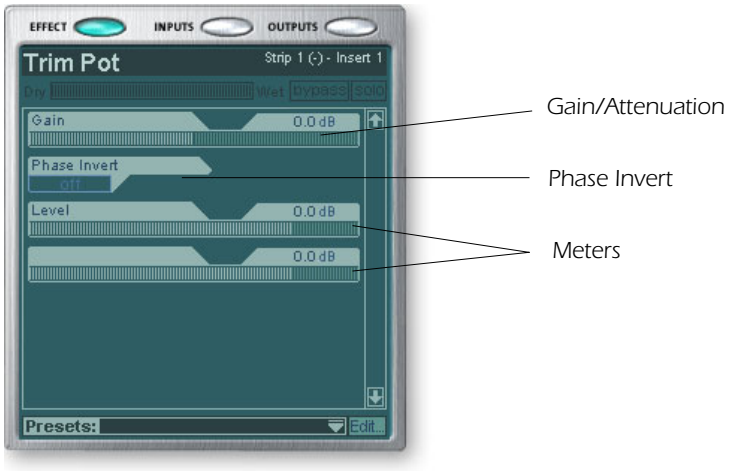
Play your input source signal while watching the insert meter in the strip. The signal should go into the yellow area frequently, but never into the red. Adjust the level of your source until you have a good level.

Digital audio has NO headroom past 0dBFS (FS = Full Scale) and will “hard clip” if the signal exceeds 0dB. Hard clipping sounds bad and will ruin your recording. Hard clipping occurs because at 0dBFS, all 24 bits are turned on and the A-D cannot measure any higher level. Analog tape, unlike digital, can be driven past 0dB, although with some degradation of the signal.

The Digital Audio System includes Insert “Trim Pot” controls, but they adjust the signal level after the signal has been digitized and will not recover any lost resolution. It’s far better to set the input level correctly in the first place. Trim Pots can be used in emergency situations if there's no other way to get a hot signal in, but they were designed to adjust levels feeding effect plug-ins.

Trim Pot Insert

The Trim Pot Insert allows you to adjust the level of a signal in an insert location. The trim pot provides up to ±30dB of gain or attenuation and a phase inverter. The trim pot also has a built-in stereo peak meter after the control.



You might use a trim pot to boost or attenuate a signal send or return from an external effect, or use it to drive an effect device. Certain effects such as the Compressor, Distortion, or Auto-Wah are very level dependent and like to see a good strong input signal. If you are working with a weak signal, you can improve the performance of these effects inserting a trim pot and boosting the gain.

Trim pots can be used to boost the level of analog line level inputs, but it's much better to boost the signal level before the A/D converters in order to get maximum resolution and signal-to-noise ratio.

The phase invert switch inverts the polarity of the signal. It is generally used to correct for mics that are wired backwards.

Test Tone/Signal Generator Insert

The test tone/signal generator insert is a handy troubleshooting aid which outputs a calibrated sine wave, white noise or pink noise. This tool, in combination with an insert meter, allows you to accurately measure the signal gain or attenuation of an internal or external device. The test tone can also be quite handy for tuning up musical instruments.

The sine wave oscillator frequency is variable from 20Hz-20kHz. The level is variable from off to +30dB.

White noise is a mixture of all frequencies in the audio spectrum at the same average level (analogous to white light in the visible spectrum).

Pink noise provides equal power distribution per octave. (White noise has more power in the higher octaves.) Pink noise and white noise are useful as wideband sound sources.

- Musical Note Freq.

A = 440 Hz
B = 493.88 Hz
C = 523.25 Hz
D = 587.33 Hz
E = 659.26 Hz
F = 698.46 Hz
G = 783.99 Hz

Managing Your Inserts

▶ To Delete Effects from an Insert:

1. Right-Click over the Insert Effect you wish to delete. A yellow line around the insert location indicates that it is selected. A pop-up dialog box appears.
2. Select Delete Insert to remove the selected insert or select Delete All Inserts to remove all inserts.
3. The insert(s) are deleted from the insert chain.

● **Tip:** Select the Plug-in and press the **Delete** key to delete the plug-in from the strip.

▶ To Bypass an Insert:

Inserts can be bypassed if you want to temporarily hear the audio without the effect or insert. Bypass can also be used to turn off a Send Insert.

Method #1

1. Click on the Effect (in the Insert section) and select FX in the TV display.
2. Click the Bypass button.

Method #2

1. Right-Click over the Effect you want to bypass (in the Insert section). A pop-up dialog box appears.
2. Select Bypass Insert from the list of options.

▶ To Bypass All Inserts:

All Inserts in a strip can be also be bypassed with a single command.

1. Right-Click over the Effect you want to bypass (in the Insert section). A pop-up dialog box appears.
2. Select Bypass All Inserts from the list of options.

▶ To Solo an Insert:

Inserts can also be soloed. Solo bypasses all the other inserts in the strip and allows you to hear only the soloed effect. This feature is very useful when adjusting the effect parameters.

Method #1

1. Click on the Effect (in the Insert section) and select FX in the TV display.
2. Click the Solo button.

Method #2

1. Right-Click over the Effect you want to Solo (in the Insert section). A pop-up dialog box appears.
2. Select Solo Insert from the list of options.

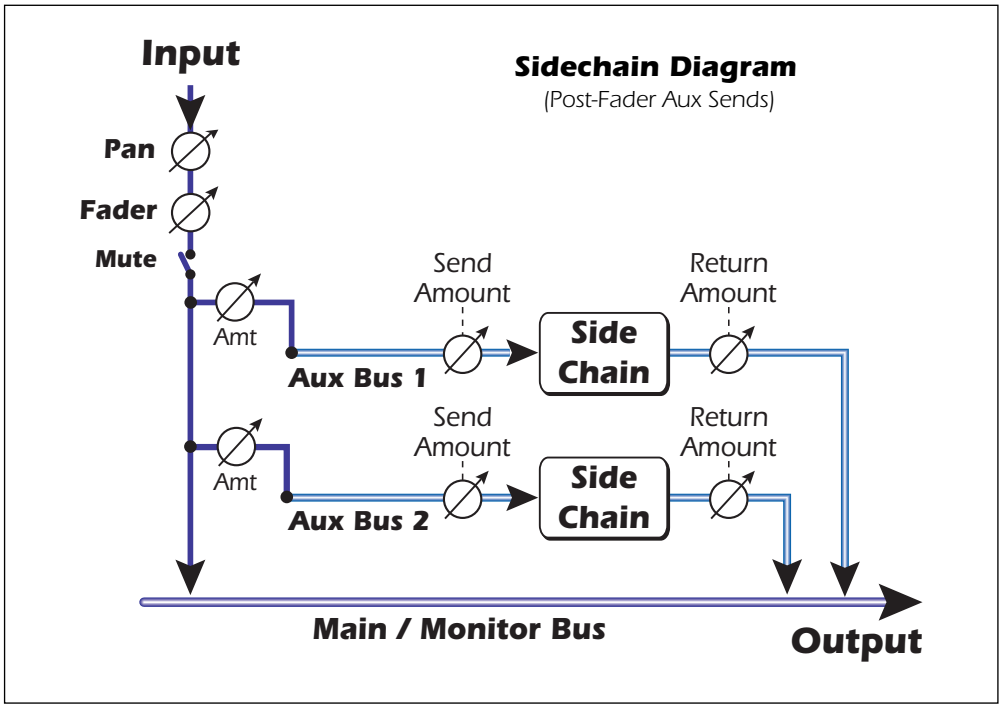
Aux Section

The Auxiliary Sends tap the signal from the channel strips and sum them together before sending the mix to the Auxiliary Effects section. In a traditional mixing console, aux sends are used to send part of the signal to outboard effect devices, then return the effected signal back into the mix using the effect returns. This is called a sidechain routing because the aux signal takes a detour through the effects before being summed back into the main mix. Sidechain effects are usually effects that you might want applied to several channels, such as reverb.

Incidentally, the wet/dry mix of effects in the Aux Sends should normally be set to 100% wet. This is because you will be adjusting the effect amount using the Aux Return control. If you have more than one effect in an Aux Bus, ignore the preceding advice as the wet/dry controls can be used to mix the amounts of your multiple effects.

The Aux 1 & 2 buses can also be used as additional submix output buses just like the main output. Simply drop an ASIO or External Send Insert into the chain and the stereo bus is sent. Turn down the Return Amount if you don't want the submix to be combined into the main mix.

Aux Send and Return values can also be changed by typing directly into the displays.



Submixing

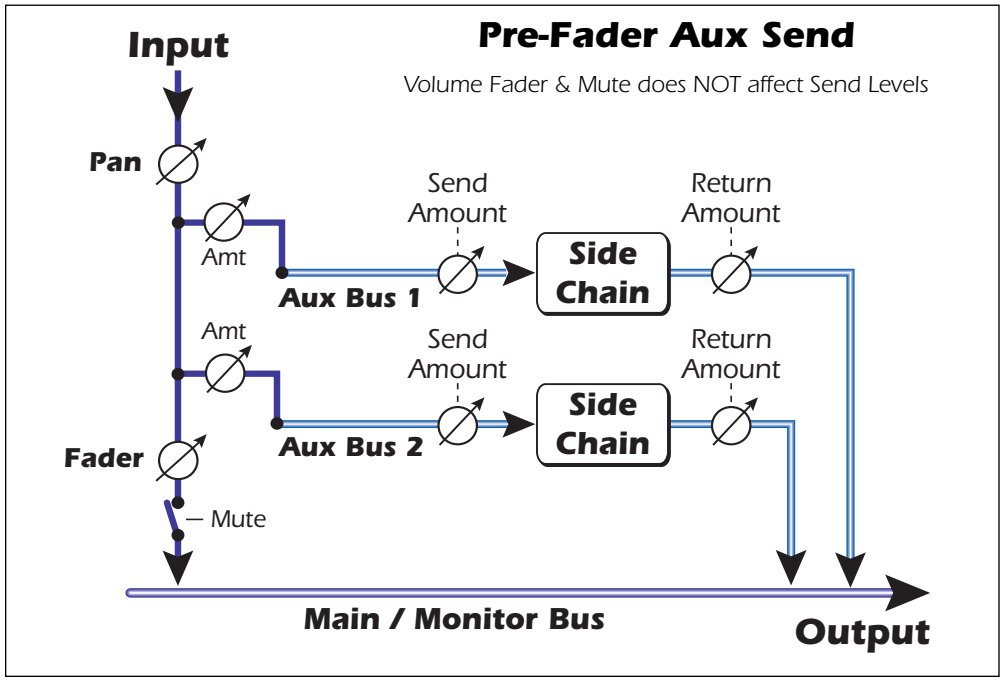
You can think of the Aux Sends as two extra mixing buses because that's exactly what they are. These two mixes can be routed anywhere, such as to a physical output or an ASIO pair. You could route one of the Aux buses to the Monitor out to create a monitor mix while sending the main mix off to your audio recording software.

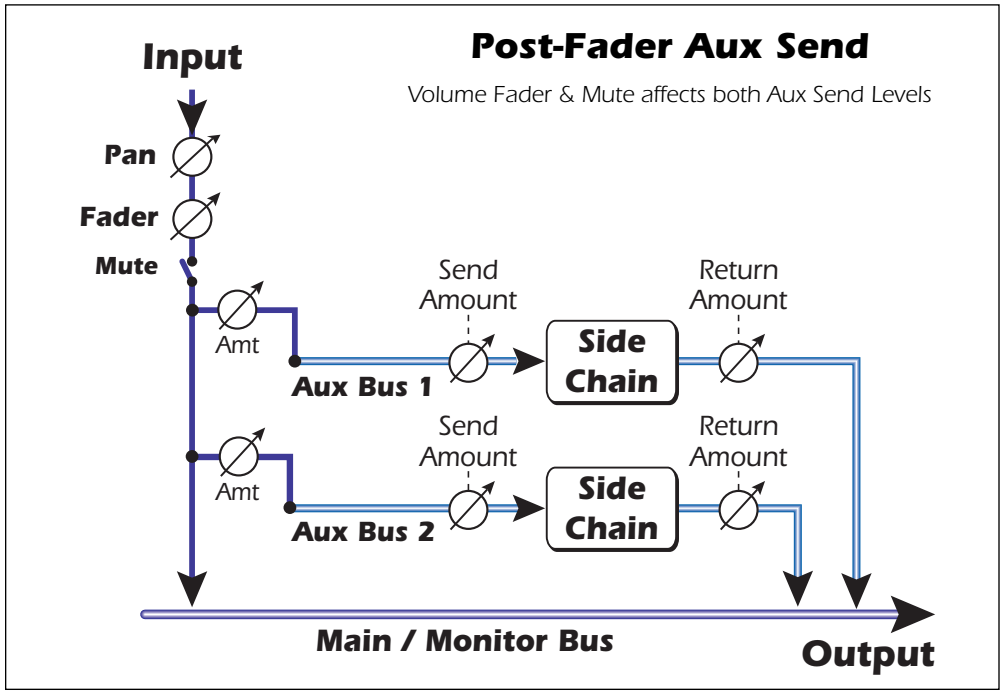
Pre or Post Fader Aux Sends

When you create a New Mixer Strip you have the option to place both Aux Sends after the channel volume fader and mute control or you can place them before the fader and mute. Post-Fader turns down the send level as you lower the volume of the strip. With Pre-Fader selected, you may still hear the effected signal returning from one of the Aux Buses with the volume fader turned down.

With the Pre-Fader box selected, the Aux Send levels are completely unaffected by the Level Fader and Mute settings. The Pre-Fader setting allows you to create two completely different mixes using the Aux Buses since the signal levels of this mix won't be affected by the fader settings.

▼ In order to change a strip from pre-fader to post-fader or vice-versa, you have to delete the strip and create a new one.





Level, Pan, Solo & Mute Controls

Pan Controls

Aux Send Amount Controls

Level Control

Mute & Solo Buttons

Scribble Strip

The **Pan** control comes before the Level Control and Aux Sends in the signal flow. On stereo strips we use an unconventional pan section with two pan pots – one for the left part of the signal and one for the right part of the signal. This feature allows you to independently position both sides of the stereo signal. A conventional stereo balance control only allows you to turn down one side or the other.

The **Mute** button does just what you would expect—press the button and the sound from that channel is cut off. Pressing the Solo button while the Mute button is pressed allows you to hear the channel until solo is turned off.

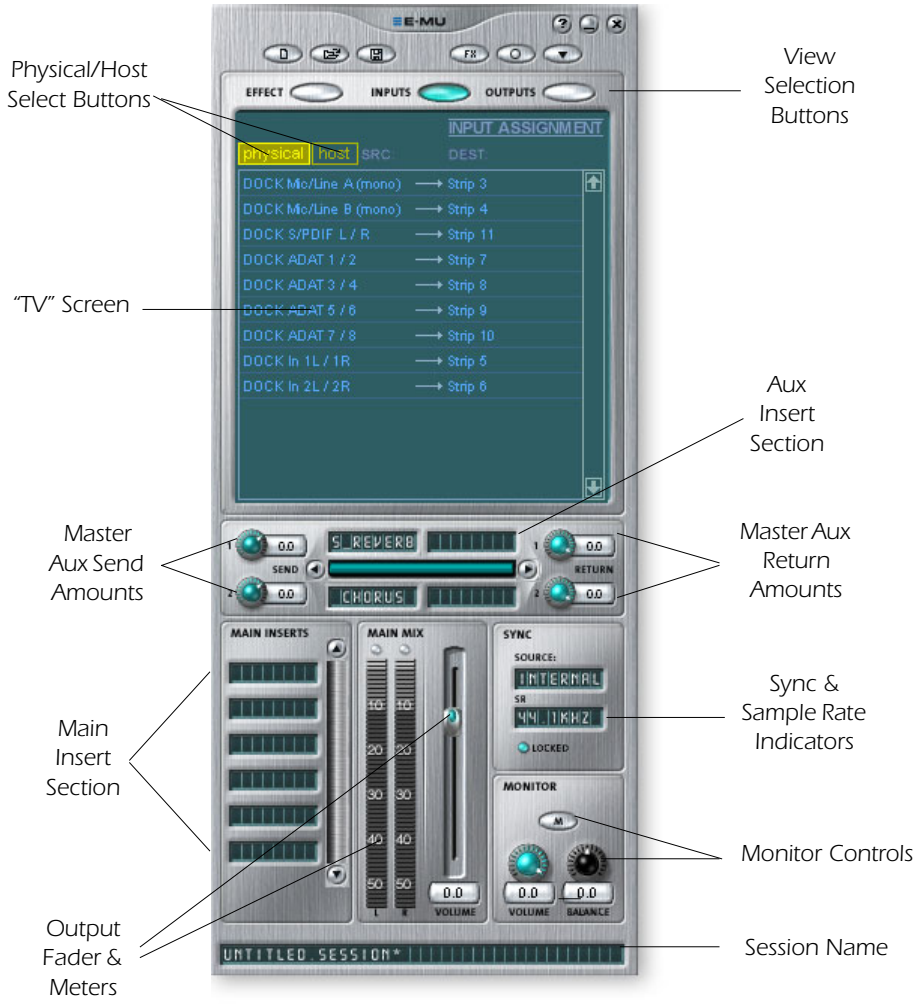
The **Solo** button allows you to listen to only that channel while muting the rest of the mixer's output. If multiple solo buttons are pressed, you will hear all soloed channels and the non-soloed channels will all be muted.

The mute status is remembered if a muted channel is soloed. When the channel solo is turned off, the channel reverts to being muted.

The **Level Control** for the strip is an attenuation control that can also provide up to +12dB of gain. 0db is the unity gain setting. You can also type numeric values into the displays to set the level.

At the very bottom is the **Scribble Strip** text area, into which you can type any short piece of text, thus naming the strip, i.e. "vocals", "bass", "drums" and so on.

Main Section



The main section contains all controls for controlling the main mix elements as well as a "TV screen" for viewing the parameters of the current selected insert.

The three buttons across the top of the main section select what is shown on the TV display. Input and output routings are graphically displayed. When an insert is selected (by clicking on the insert), the screen shows the available parameters for the currently selected insert.

Below the TV screen is the Aux Bus section where effects, effects chains or other inserts can be assigned to the two aux buses. Send and return levels can be individually controlled for each of the two Aux Buses.

The Aux 1 and Aux 2 buses are fed by the two Aux Sends on each mixer strip. The Master Send Level control on Aux bus 1 and 2 can be used to attenuate or boost the signal going into the Auxiliary Inserts. There is also a Master Return Level to control the amount of the effected signal that will be returned into the main mix.

The Main Bus can also have a chain of effects inserted. (You might put an EQ effect here to equalize your entire mix or add an ASIO or WAVE send to record the mix.) Note that the Main Output level control comes before the Monitor Level so that you can control the monitor level without affecting the level of your recording mix or main mix. There is a stereo peak meter that indicates the signal strength for the main mix.

The Monitor section has a volume, balance, and a mute control to cut off the monitor output.

TV Screen & Selectors

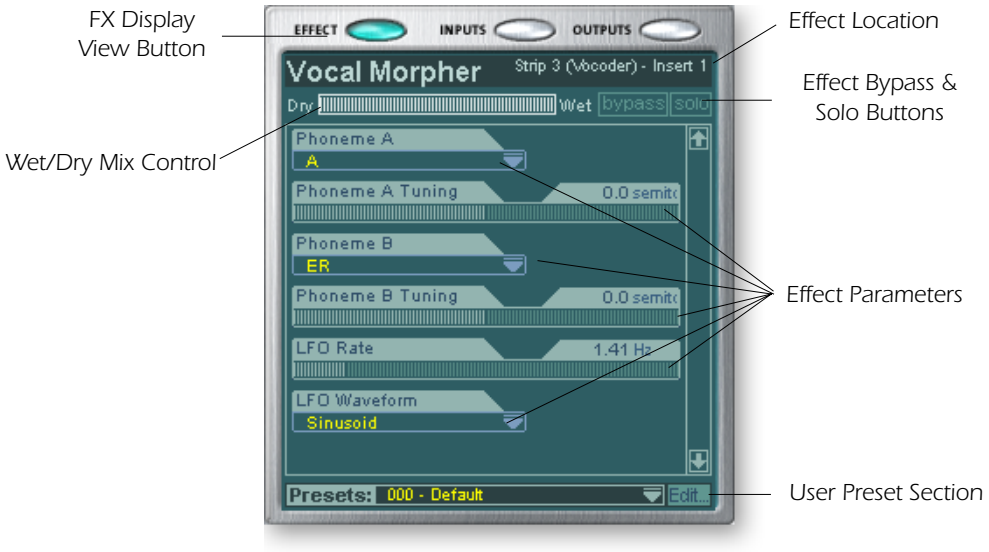
The "TV screen" at the top of the main section is a multi-function display and control center for the input and output routings and effect controls. The three buttons at the top of the display select the current function of the display—Effect, Inputs or Outputs.

Effect

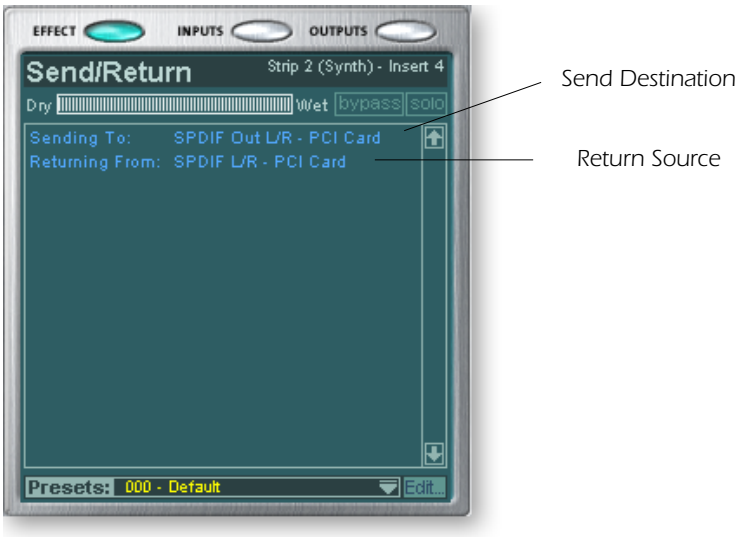
Select the **Effect** display view in the main section, then click on an Effect Insert to display the effect parameters. If an insert effect is not selected, the display will read "No Insert".

Most effects have a wet/dry mix parameter to control the ratio of effect-to-plain signal. The wet/dry setting is stored with the FX preset. The parameter set varies with the type of effect. See "Core Effects Descriptions" for detailed information about the individual effects.

■ **Note:** *Effects have to be placed into an insert location before you can program them.*



When a Send or a Send/Return insert is selected with the FX display enabled, the TV screen shows you where the Send is going and where the Return is coming from. The buttons at the top of the display allow you to bypass or solo the Send/Return insert.



Input

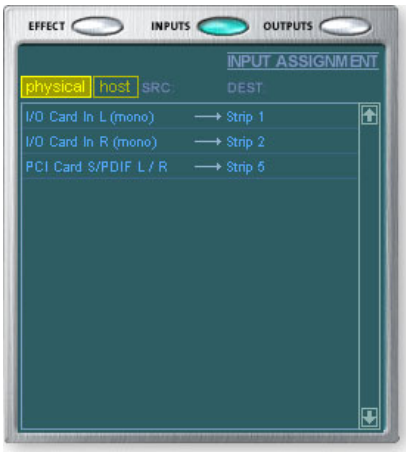
Selecting the **Input** display view shows a graphic representation of the PatchMix DSP Mixer inputs. This screen is only a display unlike the Effects and Outputs screen, which allow you to make routing changes. Input routing changes are made by adding mixer strips. See Mixer Strip Creation.

The input routings are divided into two categories: Physical Inputs and Host Inputs. Select either category by clicking on the Physical or Host button. Clicking on any of the input routings in the TV display highlights the corresponding mixer strip.

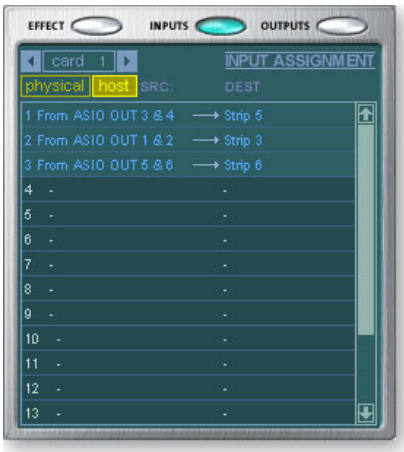
● *The Input and Output displays make it much easier to understand the signal routings of a complex mixer setup.*

● ***Tip:*** *Clicking on any of the input routings in the TV display highlights the corresponding mixer strip.*

Physical Input Display



Host Input Display

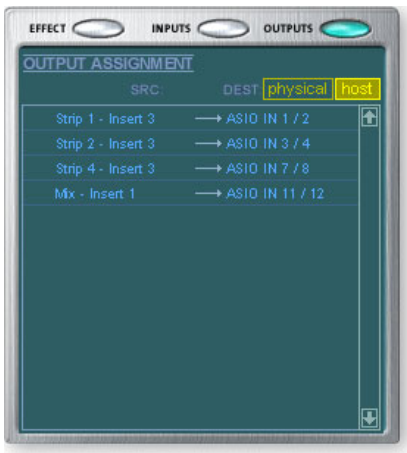


Output

Selecting the **Output** display view shows a graphic representation of the PatchMix DSP Mixer outputs. The output routings are divided into two categories: Physical Outputs and Host Outputs. Select either category by clicking on the Physical or Host button.

Physical Output Display

Host Output Display


EFFECT
INPUTS
OUTPUTS
OUTPUT ASSIGNMENT
SRC:
DEST
physical
host
Strip 1 - Insert 3 → ASIO IN 1 / 2
Strip 2 - Insert 3 → ASIO IN 3 / 4
Strip 4 - Insert 3 → ASIO IN 7 / 8
Mix - Insert 1 → ASIO IN 11 / 12

The Host Output display shows all the Insert Routings in addition to the Main Mix and Monitor out routings. Click on the desired row to make or break a physical output connection.

The **Physical Output** screen displays and allows you to connect the Main and Monitor outputs of the mixer to "physical" analog or digital outputs. Click on the box in the mix or monitor area to make (or break) a connection.

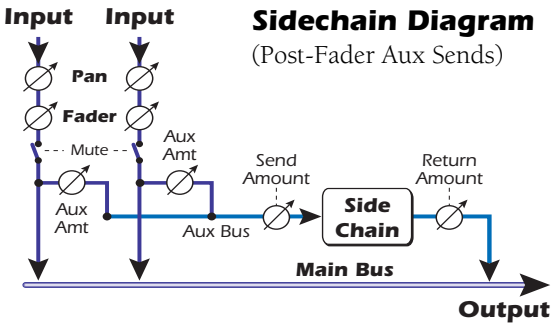
The **Host Output** screen displays and allows you to view the Host (ASIO or WAVE) outputs of the mixer. See "Insert Section" for information on how to connect the inserts.

Auxiliary Effects & Returns

The section immediately below the TV Screen is where you assign the Auxiliary Effects. In a traditional mixing console, auxiliary effects sends are used to send part of the signal to outboard effect devices, then return the effected signal back into the mix using the effect returns. This is called a sidechain routing because the aux signal takes a detour through the effects before being summed back into the main mix.

Sidechain effects are usually effects that you might want applied to several channels, such as reverb. Effects such as EQ and compressors are usually NOT used as sidechain effects because they can cause unpredictable results when returned to the main bus.

● *The Wet/Dry mix setting in the effect should normally be set to 100% when the effect is inserted as a sidechain effect. This is because the Aux Return Amount will control the wet/dry mix.*



Sidechain Diagram
(Post-Fader Aux Sends)

You can also use the Auxiliary Sends as two extra mix buses. By turning the Aux Return amount all the way down and dropping an Insert Send into the chain, you can send the Auxiliary bus to any output you wish. See "Insert Section" for more information.,

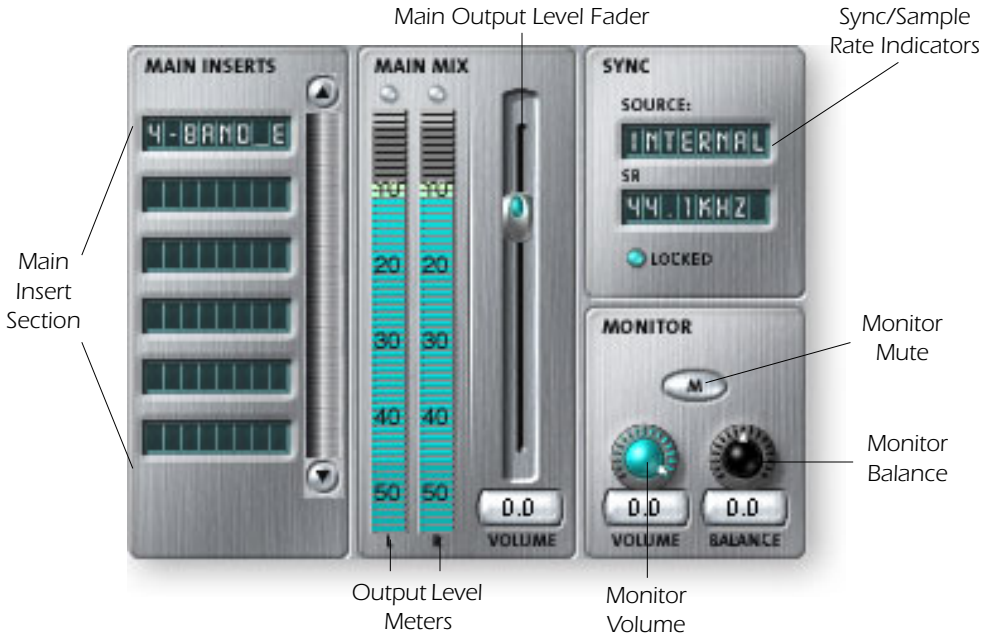
Sync/Sample Rate Indicators

The Sync/Sample rate Indicators show the current session's sample rate and whether it is internal or slaving to an external source. The LEDs indicate which sample rate is currently in effect. If an external source is being used, the Source display reads "EXTERNAL".

When slaving to an external master source, the clock may drift slightly or change dramatically (i.e. abrupt sample rate change or unplugging of physical master source). PatchMix DSP is tolerant to minor drifting within the supported rates of 44.1k, 48k, 88.2k, 96k, 176.4k, but if the sample rate drifts out of this range the "LOCKED" LED will extinguish.

If the external clock source makes a radical sample rate change from the lower rates of 44.1k/48k to a higher rate or between the rates of 88k/96k and 176k/192kHz, the hardware automatically switches to internal 48kHz clock until the proper external clock is restored. The "LOCKED" LED will be off and the two units are NOT synchronized. Always check the "LOCKED" LED when using an external clock source.

Output Section



Main Inserts

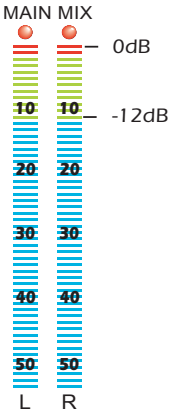
The main inserts allow you to apply effects to the main stereo signal coming out of the mixer (both mains and monitor). You might want to apply EQ or a compressor here. These inserts work just like the other effect locations—just drag and drop effects from the palette or right-click and add Sends, Sends/Returns. etc. Refer to the Mixer Block Diagram

Main Output Fader

The main output fader controls the level of the main output (and the Monitor output as well since it is downstream from this control). The normal setting for this control is at unity or 0dB, but the control allows you to add up to +12dB of gain. High output levels may cause clipping on outboard amplifiers or other equipment.

Output Level Meters

This stereo bar-graph meter reflects the digital level at the output of the mixer. The topmost red bar represents 0 dB or a full-scale digital signal. The peaks hold for a moment so that short transients can be monitored. Each bar = 1dB.



Monitor Output Level

This control adjusts the monitor output level. Keep in mind that since the monitor level control comes after the Main Output Fader, nothing will be heard from your monitors if the main level is turned down.

● **Tip:** The System Volume Control on your Mac or PC can be used to control the Monitor Output Level on PatchMix.

Monitor Balance Control

This control sets the relative volume of the stereo monitor outputs and works just like the balance control on your home music system. This control is primarily used to make the volume from each speaker sound equal if you are not sitting exactly in the center of the two speakers.

Monitor Output Mute

This button completely cuts off the monitor output and provides a convenient way to instantly kill all sound without having to re-adjust the monitor level later. When the telephone rings, just hit the monitor mute to cut the noise.

5 - Effects

Overview

PatchMix DSP comes complete with a host of great core DSP effects including Compressors, Delays, Choruses, Flangers and Reverb. Each 32-bit effect has various parameters for editing, as well as factory presets. You can also create and save as many of your own effect presets as you wish.

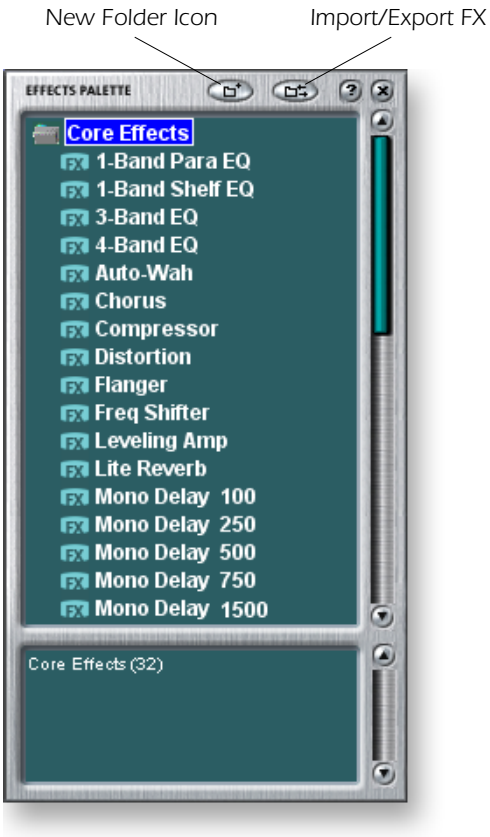
Since the effects are implemented in hardware, they don't place any load on your host computer. This allows your valuable CPU cycles to be used for other applications or software plug-ins. The effects are only available at the 44.1 and 48kHz sample rates.

There is a finite limit to how many effects you can use at the same time. As you use up the PatchMix DSP resources, certain effects will appear "grayed out" and cannot be added to the mixer. Complex effects such as reverb use more DSP resources than say, a 1-Band EQ. If you continue to add effects, all of the DSP resources will eventually be used up.

● *Saving a session "defragments" the effect/DSP resources. If you have used all your effects and need another, try saving the session.*

The Effects Palette

Click the FX button on the toolbar to bring up the Effects Palette. The Effects Palette contains two types of folders. The "Core Effects" folder contains the effect algorithms themselves. This folder cannot be modified. The other folders contain "Effects Chains", consisting of two or more effects grouped together. You can also add, delete, or modify Effects Chains and the folders that contain them. For more information on Effects Chains, see More Information.



Effect Categories

Core Effects

Multi-Effects
Distortion Lo-fi
Drums & Percussion
Environment
Equalization
Guitar
Morpher
Multi Effects
Reverb
Synths & Keys
Vocal

▶ To Select an Effect

1. Click the FX button to bring up the Effects Palette. The effect palette contains numerous folders containing effects presets. Click on any folder to open it.
2. Select the effect you wish to use by clicking on it with the left mouse button and while continuing to hold the mouse button, drag the effect into the desired location on the PatchMix DSP mixer screen and release the mouse button. Multi effects contain several effects along with their parameter settings.
3. If you want to change the order of effects, simply Left-click and drag the effect to the desired location. Drag the effect to the area above or below the final destination and release the mouse button to move the effect.

● *The order of effects in a chain can have a big effect on the sound.*

● *This icon will appear when you drag an effect to a new location.*

▶ To Edit an Effect

1. Click on the Insert Location containing the effect you wish to edit. The effect controls now appear on the TV screen.
2. Edit the effect parameters as desired.

▶ To Delete an Effect

1. Right-click on the Insert location containing the effect you wish to delete and a pop-up list appears.
2. Select “Delete Insert(s)” from the top of the list. The effect will be deleted.

FX Insert Chains

FX Insert Chains can be used to save one or several effects and their settings into a single multi-effect. When an effects chain is selected and placed into an insert location, all the effects with control settings are copied as a single entity. Once dropped into an insert location, the effects are totally separate just as if you had placed them individually.

▶ To Save FX Insert Chains

● *Trim pots, peak meters and test tone generators will also be included in the FX chain.*

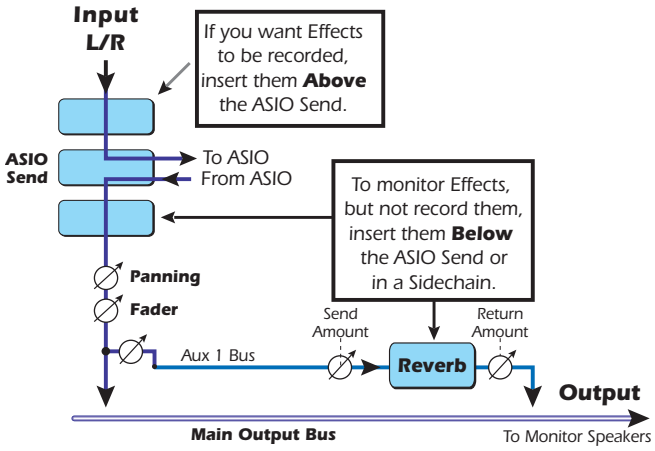
1. Select one or more effects and place them into any insert location in the mixer.
2. Set the effect parameters the way you want them, including wet/dry mix settings.
3. Right-click to bring up the list of options.
4. Select “Save FX Insert Chain”. The New FX preset dialog box appears.
5. Select a category folder where your preset will be placed, and enter a new preset name for your FX Chain.

6. Select a folder where your new preset will be placed, then type in a new preset name and click OK. Your preset is now saved.

The Order of Effects

PatchMix DSP allows you to record your tracks without effects (dry) and monitor with effects enabled (wet). It works like this: If the effect is inserted *BEFORE* the ASIO send in the signal path, it will get recorded; if the effect is inserted *AFTER* the ASIO send, it will not be recorded.

Recording dry allows you to hear your performance with effect (to get the proper feel), but gives you the flexibility to add or modify effects later during mixdown. This way if you don't like the way the effect sounds, you can change or modify the effect without having to perform the part again.



Creating, Renaming & Deleting Categories or Presets

There are several utilities to help you organize your effects presets.

▶ To Create a New Preset Category

You can create your own category folders to help organize your effects presets.

1. Left-click on the New Folder icon at the top of the Effects Palette. A pop-up dialog box appears asking you to "Enter the Name of the New Category."
2. Type in a name for your new folder.
3. Click OK to create a new folder or Cancel to cancel the operation.

▶ To Delete an Effect Category or Preset

1. Right-click on the category folder you wish to delete. A pop-up selection box appears, warning you that this action will delete all presets in the folder.
2. Select "Delete Category". A popup selection box appears.
3. Click OK to delete the folder or Cancel to cancel the operation.

▶ To Rename an FX Category

1. Right-click on the category folder you wish to rename. A pop-up selection box appears.
2. Select "Rename Category". A pop-up dialog box appears, asking you to "Enter New Category Name."
3. Click OK to rename the folder or Cancel to cancel the operation.

88kHz, 96kHz, 176kHz & 192kHz Operation

When operating at 88kHz 96kHz, 176kHz and 192kHz sample rates, the effect processors are completely disabled. However, the Inserts, Send/Returns, Meters, Trim Controls, Test Tones and ASIO Direct Monitoring ARE fully functional.

Importing and Exporting Core FX Presets and FX Insert Chains

These utilities make it easy to import or export your FX Presets and FX Insert Chains. You can share presets with your friends or download new presets from the Internet.

▶ To Import Core FX Presets

This option imports complete folders of Core FX presets into the E-MU PatchMix DSP folder (normally located here: "C:\Program Files\Creative Professional\E-MU PatchMix DSP\Core Effects"). If the name of an imported FX preset exactly matches a preset you already have, a number will be appended to end of the imported preset name.

1. Click the **Import/Export FX Library** button from the FX Palette.
2. Select **Import FX Library**. The "Browse for Folder" window appears.
3. **Choose the folder** where the Core FX presets you wish to import are located.
4. The selected folder of Core FX presets will be copied into the **Core Effects** folder of PatchMix DSP.

▶ To Import FX Category Folders

This option imports complete category folders of FX Chains into the E-MU PatchMix DSP folder (normally located here: "C:\Program Files\Creative Professional\E-MU PatchMix DSP\Effect Presets"). If the name of an imported FX preset exactly matches a preset you already have, a number will be appended to end of the imported preset name.

1. Click the **Import/Export FX Library** button from the FX Palette.
2. Select **Import FX Category**. The "Browse for Folder" window appears.
3. **Choose the folder** where the FX Chains you wish to import are located.
4. The selected folder of FX Chains will be copied into the **Effect Presets** folder of PatchMix DSP.

▶ To Export your Core FX Presets

This option exports your Core FX presets to a folder of your choice.

1. Click the **Import/Export FX Library** button from the FX Palette.
2. Select **Export FX Library**. The "Browse for Folder" window appears.
3. Choose a destination location for the Core FX presets, then press **OK**.
4. The Core FX presets will be copied to the selected destination.

▶ To Export your FX Category Folders

This option exports a single category of FX chains to a folder of your choice.

1. Click the **Import/Export FX Library** button from the FX Palette.
2. Select **Export FX Category**. A pop-up dialog box appears asking you to "Choose the FX Category to be exported".
3. **Choose the desired FX Category** to export. Press **OK** to continue or **Cancel** to cancel the operation.
4. The "Browse for Folder" window appears. **Choose a destination** location for the Core FX presets, then press **OK**.
5. The FX Chains will be copied to the selected destination.

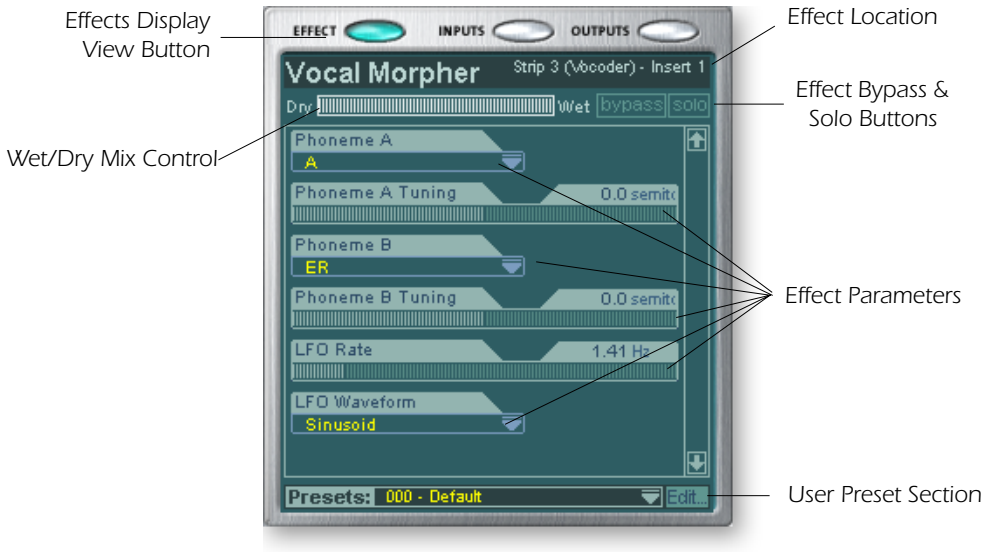
FX Edit Screen

Click on an FX Insert to display the parameters for that effect. If an insert effect is not selected, the FX display will read “No Insert”.

Most effects have a wet/dry mix parameter to control the ratio of effect-to-plain signal. The wet/dry setting is stored with the FX preset. The effect parameters vary with the type of effect. Generally if an effect is placed in an Aux Send, the wet/dry mix in the effect should be set to 100% wet since the Aux Return amount controls how much effect is applied.

The User Preset section is located at the bottom of the FX Edit screen. User presets are variations of the main effect and can be edited, deleted, renamed or overwritten as you wish.

■ **Note:** Effects have to be placed into an insert location before you can program them.



▶ To Bypass an Insert:

Inserts can be bypassed if you want to temporarily hear the audio without the effect or insert. Bypass can also be used to turn off a Send Insert.

Method #1

1. Click on the Insert (in the Insert section).
2. Click the Bypass button in the TV display.

Method #2

1. Right-click over the Insert you want to bypass (in the Insert section). A pop-up menu appears.
2. Select “Bypass Insert” from the list of options. The effect name will “gray-out” to indicate that the effect is bypassed.

▶ To Solo an Insert:

Inserts can also be soloed. Solo bypasses all the other inserts in the strip and allows you to hear only the soloed effect. This feature is very useful when adjusting the effect parameters.

Method #1

1. Click on the Insert (in the Insert section).
2. Click the Solo button in the TV display.

Method #2

1. Right-click over the Insert Effect you want to Solo (in the Insert section). A pop-up menu appears.
2. Select "Solo Insert" from the list of options. The other Insert Effect names in the strip will "gray-out" to indicate that they are bypassed.

▶ To Bypass ALL

All the inserts in a strip can be bypassed with a single command.

1. Right-click over any Effect in the Insert section. A pop-up menu appears.
2. Select "Bypass All Inserts" from the list of options. All the insert names will "gray-out" to indicate that they are bypassed.

▶ To Un-Bypass ALL

All the inserts in a strip can also be un-bypassed with a single command. This command works even if only some of the effects are bypassed.

1. Right-click over any Effect in the Insert section. A pop-up menu appears.
2. Select "Un-Bypass All Inserts" from the list of options. All the insert names will light to indicate that they are active.

User Preset Section

Each core effect has a set of User Presets, that you can use to store your favorite effect parameter settings. We've included a good collection of user presets to get you started. The user presets are accessed from the bar at the bottom of the TV screen. The user preset edit menu allows you to select stored presets, create new presets, rename or delete existing presets, or overwrite existing presets with your modified settings. User presets stay with the Mixer application regardless of which Session is open.

■ To copy or share User Presets, you must save them as FX Palette effects.

▶ To Select a User Preset

1. Select the FX display in the TV screen.
2. Select the desired insert effect, highlighting it. The effect parameters appear in the TV screen.
3. Click on the ▼ icon on the preset menu. A drop-down preset list appears.
4. Select a preset from the list.

▶ To Create a New User Preset

1. Select the FX display in the TV screen.
2. Select the desired insert effect, highlighting it. The effect parameters appear in the TV screen.
3. Click on the Edit button. A pop-up menu appears.
4. Select New. A pop-up dialog box appears asking you to name the new preset.
5. Name the preset and click OK. Your new preset is now saved.

▶ To Delete a User Preset

1. Select the user preset you wish to delete from the user preset menu.
2. Click on the Edit button. A pop-up menu appears.
3. Select Delete. A pop-up dialog box appears asking you to confirm your action.
4. Click OK to delete the preset or No or Cancel to cancel the operation.

▶ To Rename a User Preset

1. Select the user preset you wish to rename from the user preset menu.
2. Click on the Edit button. A pop-up menu appears.
3. Select Rename. A pop-up dialog box appears asking you to rename the preset.
4. Type in the new preset name, then click OK to rename the preset or Cancel to cancel the operation.

▶ To Overwrite or Save a User Preset

This operation allows you to overwrite an existing preset with a newer version.

1. Select the user preset you wish to modify from the user preset menu and make any changes you wish.
2. Click on the Edit button. A pop-up menu appears.
3. Select Overwrite/Save. The current preset will be overwritten with the new settings.

Core Effects and Effects Presets

The Core Effects cannot be removed or copied. Effect presets (stored in "C:\Program Files\Creative Professional\Digital Audio System\E-MU PatchMix DSP\Effect Presets") can be copied, e-mailed or shared like any other computer file.

Hint: Open with "NotePad" or other word processor to view and edit the name and parameters.

WDM Recording and Playback Behavior

WDM record and playback is now supported at all PatchMix sample rates. The behavior of the driver with respect to PatchMix sample rate is described below.

When PatchMix and the WDM audio content (.WAV file format, record and playback settings in WaveLab. etc.) are both running at the same sample rate, and when a Wave strip or send is present in the PatchMix mixer configuration, WDM audio will be played or recorded "bit accurate" without sample rate conversion or bit truncation.

When running PatchMix at 44kHz/48kHz, if there is a mismatch between the WDM record or playback audio content and the PatchMix sample rate, sample rate conversion is performed, so that WDM audio will always be heard or recorded. Also, such non-native-sample-rate audio is truncated to 16-bits.

When running PatchMix at 88.2kHz/96kHz or 176.4kHz/192kHz, WDM record or playback audio content must be running at the same sample rate as PatchMix. If the sample rates are mismatched, NO AUDIO will be recorded or played back. In other words, the WDM driver does not perform sample rate conversion of any kind when PatchMix is running at 88.2kHz/96kHz or 176.4kHz/192kHz.

List of Core Effects

Stereo Reverb	Rotary	Mono Delay 250
Lite Reverb	Phase Shifter	Mono Delay 500
RFX Compressor	Frequency Shifter	Mono Delay 750
Compressor	Auto-Wah	Mono Delay 1500
Reshaper	Vocal Morpher	Mono Delay 3000
Gate	1-Band Para EQ	Stereo Delay 100
Leveling Amp	1-Band Shelf EQ	Stereo Delay 250
Chorus	3-Band EQ	Stereo Delay 550
Flanger	4-Band EQ	Stereo Delay 750
Distortion	Multimode EQ	Stereo Delay 1500
Speaker Sim	Mono Delay 100	

DSP Resource Usage

There are two main factors which determine the total number of effects available for use at any given time: Tank Memory and DSP Instructions. Using too much of either resource will cause effects to be unavailable (grayed out) in the FX menu. In addition, the strips themselves use DSP Instructions, so only create strips that you actually need.

Tank memory is the memory used by delay-based effects such as reverb and digital delays. All the reverbs and delays aside from the Mono Delay 100 and Stereo Delay 100 use varying amounts of tank memory.

The DSP instructions are used by all the effects. Effects with multiple stages, such as multi-band EQs or the speaker simulator use more DSP instructions than a 1-Band EQ.

Tank memory tends to get used first, and so we've provided many delay line effects to allow maximum conservation of this precious resource. Use only the longest delay you actually need.

● **Tip:** *Saving a session "defragments" the effect/DSP resources. If you have used all your effects and need another, try saving the session.*

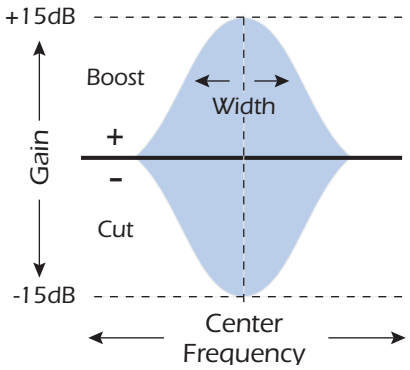
The chart below shows three possible effects combinations. These were created by using up the reverb resources first. Even more simultaneous effects are possible if fewer reverbs and shorter delays are used.

Examples of Effects Usage *(with a WAVE, ASIO Return & 2 Inputs)*

Example 1	**No.**	**Example 2**	**No.**	**Example 3**	**No.**
Stereo Reverb	2	Lite Reverb	5	Stereo Reverb	1
4-Band EQ	4	3-Band EQ	5	Lite Reverb	2
3-Band EQ	2	1-Band EQ	4	Stereo Delay 1500	1
1-Band EQ	6	Compressor	1	Mono Delay 250	1
Compressor	6	Mono Delay 1500	1	Compressor	6
Chorus	1	Mono Delay 250	1	Chorus	2
Mono Delay 1500	1	Auto-Wah	1	Flanger	2
				4-Band EQ	3
				3-Band EQ	3
Total Effects	22	**Total Effects**	18	**Total Effects**	21

Core Effects Descriptions

1-Band Para EQ

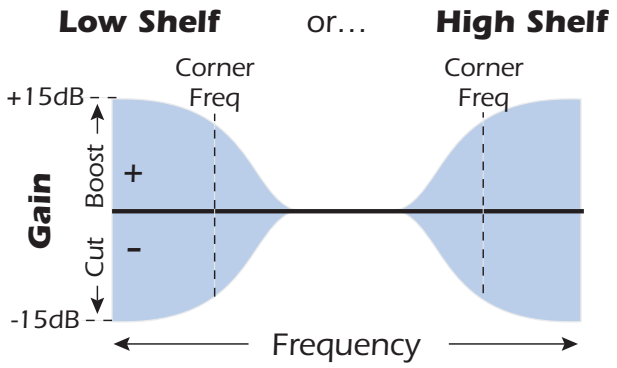


This single band parametric equalizer is useful when you just want to boost or cut a single range of frequencies. For example, if you just want to brighten up the lead vocal a bit, you might choose this EQ. This EQ offers up to ±15dB cut or boost.

Parameter	Description
Gain	Sets the amount of cut (-) or boost (+) of the selected frequency band. Range: -15dB to +15dB
Center Frequency	Sets the range of frequencies to be cut or boosted with the Gain control. Range: 80Hz to 16kHz
Bandwidth	Sets the width of the frequency range for the Center Frequency band that will be cut or boosted by the Gain control. Range: 1semitone to 36 semitones

1-Band Shelf EQ

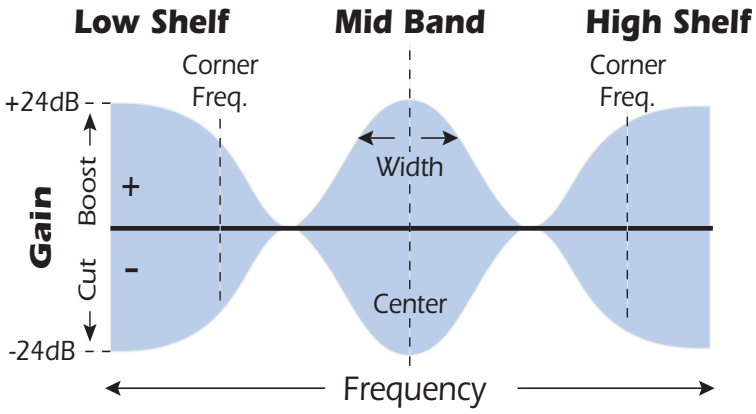
This single band shelving equalizer is useful when you just want to boost or cut a single range of frequencies at the high or low end of the spectrum. For example, if you just want to add a little more bass, there's no need to waste a 3-band EQ. Just choose low shelf, then adjust the gain and frequency. This EQ offers up to ±15dB cut or boost.



Parameter	Description
Shelf Type	Allows you to choose either low shelving or high shelving EQ.
Gain	Sets the amount of cut (-) or boost (+) of the shelf. Range: -15dB to +15dB
Corner Frequency	Sets the frequency where the signal begins getting cut or boosted with the Gain control. Range: -15dB to +15dB

3-Band EQ

This versatile equalizer provides two shelving filters at the high and low ends of the frequency range and a fully parametric band in the center. Up to ±24 dB of boost or cut is provided for each band.



■ **Note:** The Wet/Dry Mix control on an equalizer should normally be set to 100% wet or unpredictable results may occur.

▶ Setting up a Parametric EQ

1. Turn up the gain on the band you are working with. This allows you to easily hear the effect of the filter.
2. Reduce the bandwidth if you are working with a mid-band.
3. Now adjust the Center Frequency to "zero-in" on the frequencies you wish to boost or cut.
4. Set the Gain to a positive value to boost frequencies or to a negative value to cut out frequencies.
5. Widen the Bandwidth to create a more natural sound.
6. Adjust and tweak as needed.

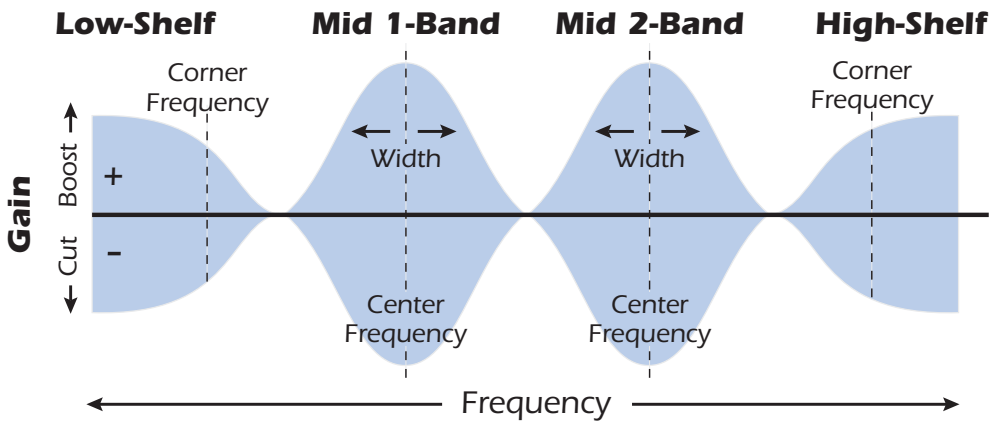
Parameter	Description
High Shelf Gain	Sets the amount of cut (-) or boost (+) of the high frequency shelf. Range: -24dB to +24dB
High Corner Freq.	Sets the frequency where the signal begins getting cut or boosted with the High Gain control. Range: 4kHz to 16kHz
Mid Gain	Sets the amount of cut (-) or boost (+) of the mid frequency band. Range: -24dB to +24dB
Mid Center Freq.	Sets the range of frequencies to be cut or boosted with the Mid Gain control. Range: 200Hz to 3kHz
Mid Bandwidth	Sets the width of the frequency range for the Mid Center Frequency band that will be cut or boosted by the Mid Gain control. Range: 1 semitone to 1 octave
Low Shelf Gain	Sets the amount of cut (-) or boost (+) of the low frequency shelf. Range: -24dB to +24dB
Low Corner Freq.	Sets the frequency where the signal begins getting cut or boosted with the Low Gain control. Range: 50Hz to 800Hz

4-Band EQ

This 4-band equalizer provides two shelving filters at the high and low ends of the frequency range and two fully parametric bands in the center. Up to ±24 dB of boost or cut is provided for each band.

Note: The Wet/Dry Mix control on an equalizer should normally be set to 100% wet or unpredictable results may occur.

For more information on setting up a parametric EQ, see page 52.



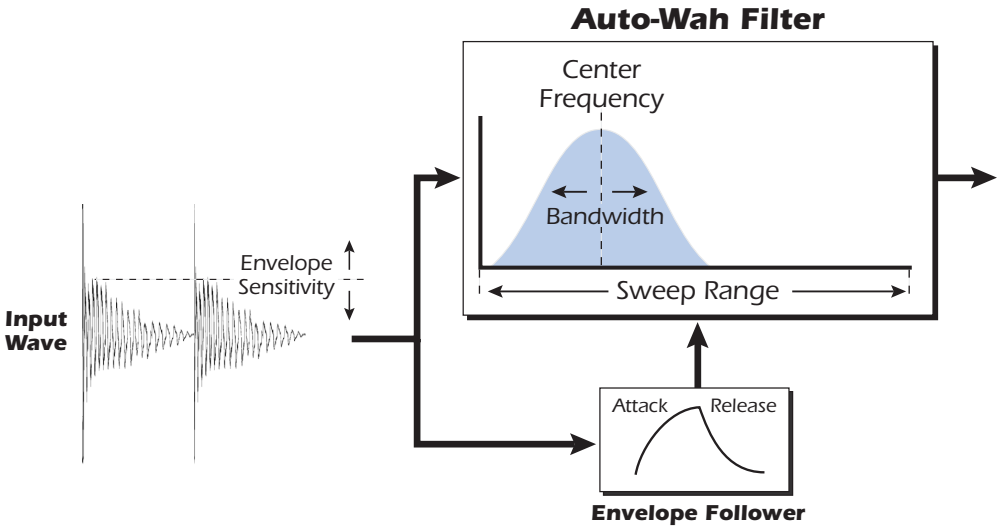
Parameter	Description
High Shelf Gain	Sets the amount of cut (-) or boost (+) of the high frequency shelf. Range: -24dB to +24dB
High Corner Freq.	Sets the frequency where the signal begins getting cut or boosted with the High Gain control. Range: 4kHz to 16kHz
Mid 2 Gain	Sets the amount of cut (-) or boost (+) of the Mid 2 Frequency band. Range: -24dB to +24dB
Mid 2 Center Freq.	Sets the range of frequencies to be cut or boosted with the Mid 2 Gain control. Range: 1kHz to 8kHz
Mid 2 Bandwidth	Sets the width of the frequency range for the Mid 2 Center Frequency band that will be cut or boosted by the Mid 2 Gain control. Range: .01 octave to 1 octave
Mid 1 Gain	Sets the amount of cut (-) or boost (+) of the Mid 1 Frequency band. Range: -24dB to +24dB
Mid 1 Center Freq.	Sets the range of frequencies to be cut or boosted with the Mid 1 Gain control. Range: 200Hz to 3kHz
Mid 1 Bandwidth	Sets the width of the frequency range for the Mid 1 Center Frequency band that will be cut or boosted by the Mid 1 Gain control. Range: .01 octave to 1 octave
Low Shelf Gain	Sets the amount of cut (-) or boost (+) of the low frequency shelf. Range: -24dB to +24dB
Low Corner Freq.	Sets the frequency where the signal begins getting cut or boosted with the Low Gain control. Range: 50Hz to 800Hz

Auto-Wah

This effect creates the sound of a guitar wah-wah pedal. The “Wah” filter sweep is automatically triggered from the amplitude envelope of the input sound. Auto-wah works well with percussive sounds such as guitar or bass.

The Auto-Wah is a bandpass filter whose frequency can be swept up or down by an envelope follower, which extracts the volume contour of the input signal. The Envelope Sensitivity setting allows you to properly set up the envelope follower to receive a wide variety of input signals. This ‘envelope’, or volume contour, controls the frequency of the bandpass filter so that it sweeps up and down with each new note. The Attack controls the rate of the note-on sweep. As the input sound fades away, the filter sweeps back at a rate determined by the Release setting.

The wah direction allows the filter to be swept either up or down in frequency. Use a higher Center Frequency setting when the wah direction is down.



Parameter	Description
Wah Direction	Allows you to sweep the wah up or down.
Env. Sensitivity	Controls how closely the wah sweep follows the input signal. Range: -12dB to +18dB
Env. Attack Time	Sets the starting rate of the “wah” sweep. Range: 0ms to 500ms
Env. Release Time	Sets the ending or release rate of the “wah” sweep. Range: 10ms to 1000ms
Sweep Range	Controls the amount of “wah” sweep. Range: 0% to 100%
Center Frequency	Sets the initial bandpass filter frequency. Range: 80Hz to 2400Hz
Bandwidth	Sets the width of the bandpass filter. Range: 1Hz to 800Hz

Chorus

An audio delay in the range of 15-20 milliseconds is too short to be an echo, but is perceived by the ear as a distinctly separate sound. If we now cyclically vary the delay time in this range,

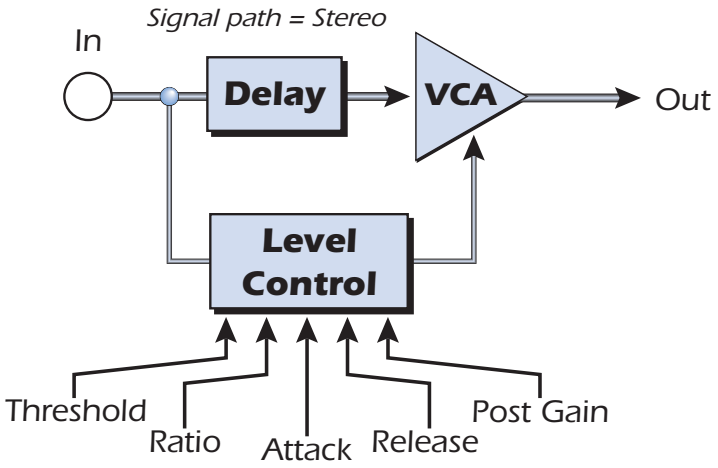
the illusion of multiple sound sources is created. A slight amount of feedback serves to increase the effect. A very slow LFO rate is usually best for a realistic effect, but a faster LFO rate can also be useful with minimal LFO depth (.2). Since this is a stereo chorus, an LFO phase parameter is included which can be used to widen the stereo image.

Parameter	Description
Delay	Sets the length of the delay. Range: 0ms to 20ms.
Feedback	Sets the amount of delayed signal that will be recirculated through the delay line. Range: 0% to 100%
LFO Rate	Sets the frequency of the low frequency oscillator. Range: .01Hz to 10Hz
LFO Depth	Sets how much the LFO affects the delay time. Increases the animation and amount of the chorus effect. Range: 0% to 100%
LFO Waveform	Selectable between Sine or Triangle wave.
LFO L/R Phase	Controls the stereo width by adjusting the phase difference of the LFO waveform between left and right channels. Range: -180° to +180°

Compressor

In its simplest form, an audio compressor is just an automatic gain control. When the volume gets too loud, the compressor automatically turns it down. Compressors are useful in musical applications because they allow you to record a "hotter" signal without overloading the recording device.

Since the compressor turns **down** the gain of the signal, you might wonder how can it make the signal level stronger. A Post Gain control allows you to boost the output gain of the compressor in order to make up for the gain reduction. The overall level is higher and only turned down when the signal level gets too loud. This level is called the *Threshold,* which just happens to be the most important control on the compressor.



Basic Controls

The three main controls of a compressor are the *Ratio* control, the *Threshold* control and the *Gain* control.

If the signal level falls below the **Threshold**, no processing will take place. Signals exceeding the Threshold will have gain reduction applied as set by the ratio control. This important control allows you to dial in the range of amplitudes you want to tame. For example, if you're trying to trim off just the loudest peaks, set the threshold so the gain reduction meter only shows compression during these peaks. One of the biggest mistakes in using a compressor is having the threshold set too low. This adds noise as the compressor will always be reducing the volume.

The **Ratio** control determines how strongly the compressor will affect the signal. The higher the ratio, the more reduction will be applied. If the ratio is high enough, *(above 10:1)* the signal will effectively be prevented from getting any louder. In this situation, the compressor will be acting as a **Limiter**, placing an upper limit on the signal level. In general, ratios from 2:1 to 6:1 are considered compression and higher ratios above 10:1 are considered limiting.

The **Post Gain** control amplifies the signal after it has been compressed to bring it back up in volume. If you don't increase the gain, the compressed signal will be much lower in volume.

Two other important controls are **Attack** and **Release**. Attack controls how quickly the gain is turned down after the signal exceeds the threshold. Release controls how fast the gain is returned to its normal setting after the signal has fallen below the threshold again. An attack setting of about 10 milliseconds will delay the onset of compression long enough to preserve the attack transients in guitar, bass or drums while allowing the sustain portion of the sound to be compressed. Longer release times are generally used to reduce the so called "pumping" effect as the compressor turns on and off. Don't make the release time too long, however, or the compressor won't have time to recover for the next pluck or hit. In general, the attack and release controls are used to smooth out the action of the compressor, but they can also be used to create special effects.

The Pre-Delay parameter lets the level detector "look into the future" up to 4 milliseconds in order to anticipate upcoming peaks in the signal. This is accomplished of course, by inserting delay into the signal path. This lookahead technique allows the use of slower attack times without missing signal peaks. This parameter is especially effective on drums and percussion.

The Input Meter allows you to monitor the strength of your input signal. Always try to boost the signal before the compressor if you can.

The Compression Meter shows the amount of gain reduction being applied. Since this meter displays how much the gain is being turned *down*, the meter moves from right to left, instead of left to right like a normal meter.

Parameter	**Description**
Threshold	Threshold sets the input signal level above which dynamic range compression takes place. Everything above the threshold will be brought down in volume. Range: -60dB to +12dB
Ratio	Sets the ratio of input signal level to output signal level, or "how much" compression will be applied. Range: 1:1 to ∞:1
Post Gain	Amplifies the signal after it has been compressed to bring up the volume. Range -60dB to +60dB
Attack Time	Controls how quickly the gain is turned down after the signal exceeds the threshold. Range .1ms to 500ms
Release Time	Controls how fast the gain is returned to its normal setting after the signal has fallen below the threshold. Range: 50ms to 3000ms

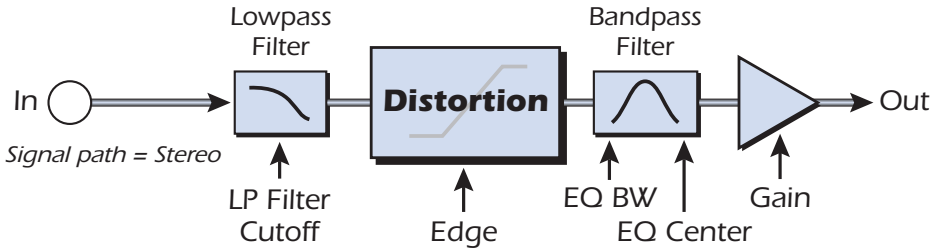
Parameter	Description
Pre-Delay	Allows the use of slower attack times without missing signal peaks. Range: 0ms to 3 ms
Input Meter	Allows you to monitor the strength of the input signal.
Gain Reduction Meter	Shows the amount of gain reduction being applied.

Distortion

Most audio processors aim to provide low distortion, but not this one! The sole purpose of this effect is to add distortion, and lots of it. This effect provides "fuzz box" style, clipping distortion which is particularly effective on guitar, bass, organs, electric pianos or whatever.

The input signal first passes through a lowpass filter. The Lowpass Filter Cutoff Frequency allows you to control the number of new harmonics that will be generated by the distortion element. The distortion element has an Edge control which controls "how much" distortion will be added. A bandpass filter follows the distortion generator. The EQ Center control lets you select a particular band of frequencies to be output. The EQ Bandwidth controls the width of the center frequency band. Finally, a gain control allows you to make up for any gain loss through the effect.

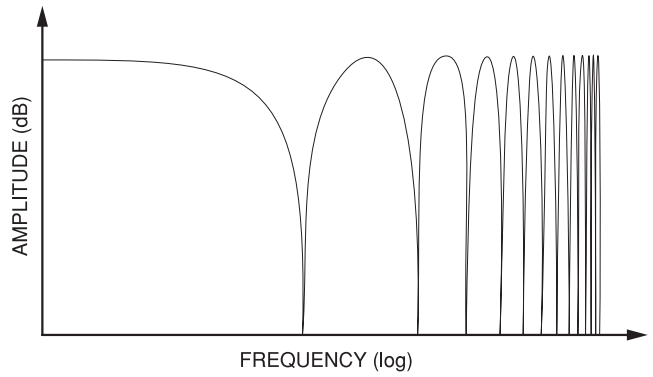
Use the Wet/Dry mix control in conjunction with the Edge control to reduce the amount of distortion, or go crazy and turn everything to 11!



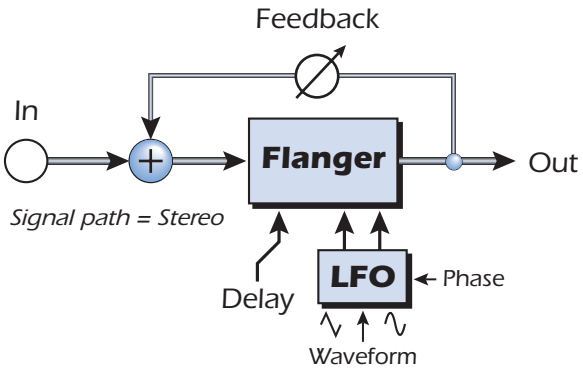
Parameter	Description
Pre EQ LP Cutoff	Controls the amount of high frequency audio admitted to the distortion. Range: 80Hz to 24kHz
Edge	Sets the amount of distortion and new harmonics generated. Range: 0-100
Gain	Sets the output volume of the effect. Range: -60dB to 0dB
Post EQ Center Freq.	Sets the frequency of the output bandpass filter. Range: 80Hz to 24kHz
Post EQ Bandwidth	Sets the width of the output bandpass filter. Range: 80Hz to 24kHz

Flanger

A flanger is a very short delay line whose output is mixed back together with the original sound. Mixing the original and delayed signals results in multiple frequency cancellations known as a comb filter. Since the flanger is a type of filter, it works best with harmonically rich sounds.



A low frequency oscillator is included to slowly change the delay time. This creates a rich, sweeping effect as the notches move up and down across the frequency range. The amount of feedback deepens the notches, intensifying the effect. You can invert the feedback signal by choosing a negative feedback value. Inverting the feedback signal creates peaks in the notch filter and deepens the effect.



Parameter	Description
Delay	Sets the initial delay of the flanger in 1/100th millisecond increments. This parameter allows you to "tune" the flanger to a specific frequency range. Range: .01ms to 4ms
Feedback	Controls how much signal is recirculated through the delay line and increases resonance. Negative values can produce intense flanging with some signals. Range 0% to 100%
LFO Rate	Sets the speed of the flanger sweep. Range: .01 Hz to 10Hz
LFO Depth	Sets how much the LFO affects the delay time. Increases the animation and amount of the flanging effect. Range 0% to 100%
LFO Waveform	Selectable between Sine or Triangle wave.
LFO L/R Phase	Controls the stereo width by adjusting the phase difference between the left and right sweeps. Range: -180° to +180°

Freq Shifter

This unusual effect is sometimes called "spectrum shifting" or "single sideband modulation." Frequency shifting shifts every harmonic in the signal by a fixed number of Hz which causes the harmonics to lose their normal relationship. The more common pitch shifter, in contrast,

preserves the harmonic relationships of the signal and so is better suited to creating "musical" harmonies.

This isn't to say that the frequency shifter can't be used musically. Small intervals of frequency shifting (1 Hz and below) can produce a wonderful, lush chorusing or phasing effect. For bizarre frequency shifting effects, simply crank up the frequency knob. Frequencies can be shifted up or down by any specified amount from .1 Hz to 24 kHz. You can also shift pitch up on one side and down on the other if you wish.

● *You can also type in exact frequencies to a resolution of 1/10 Hz.*

Comparison between Pitch and Frequency Shifting

Harmonic	**Original** (Hz)	**Pitch Shifted** (100 Hz)	**Frequency Shifted** (100 Hz)
1	200	300	300
2	400	600	500
3	600	900	700
4	800	1200	900
5	1000	1500	1100
6	1200	1800	1300
7	1400	2100	1500
8	1600	2400	1700

Parameter	**Description**
Frequency	Sets the number of Hz that will be added or subtracted with every harmonic in the signal. Range: .01Hz to 24kHz
Left Direction	Sets pitch shift up or down for the left channel.
Right Direction	Sets pitch shift up or down for the right channel.

Leveling Amp

The first compressors developed in the 1950's were based on a slow-acting optical gain cells which were able to control the signal level in a very subtle and musical way. This effect is a digital recreation of the leveling amps of yesteryear.

The leveling amp uses a large amount of "lookahead delay" to apply gentle gain reduction. Because of this delay, the leveling amp is not suitable for applications which require realtime monitoring of the signal. This smooth and gentle compressor is designed to be used in situations where delay does not pose a problem, such as mastering a mix or compressing prerecorded stereo material.

Post Gain is the only control on the leveling amp. This control is used to make up the volume lost by the compression. The Compression Ratio is fixed at about 2.5:1. If a large peak is detected, the effect will automatically increase the compression ratio to keep the audio output controlled.

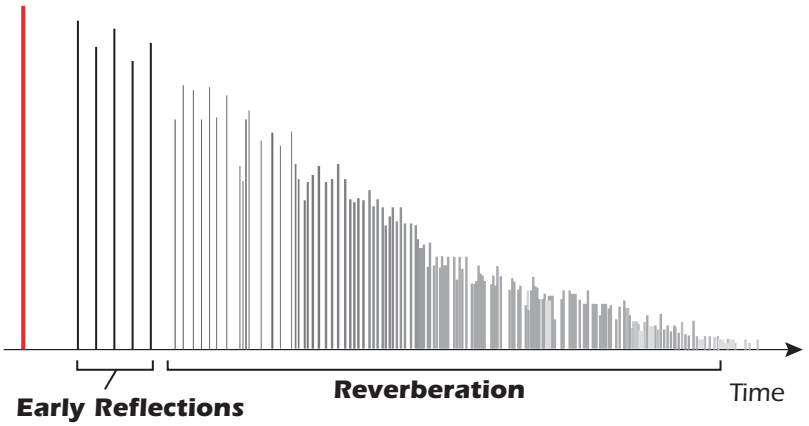
The gain reduction meter shows you how much gain reduction is being applied. Since the gain reduction meter displays how much the gain is being turned *down*, the meter moves from right to left, instead of left to right like most meters.

Post Gain	Amplifies the signal after it has been compressed to bring up the volume.

Lite Reverb

Reverberation is a simulation of a natural space such as a room or hall. The Lite Reverb algorithm is designed to simulate various rooms and reverberation plates while using fewer DSP resources than the Stereo Reverb. Up to five Lite Reverbs can be used at once.

Decay time defines the time it takes for the reflected sound from the room to decay or die away. The diagram below shows a generalized reverberation envelope.



After a short pre-delay period, the echoes from the closest walls or ceiling are heard. These first echoes, or *Early Reflections*, vary greatly depending on the type of room. Some time after the early reflection cluster ends, the actual *Reverberation* (a dense cloud of complex wall reflections) begins and decays according to the time set by the Decay Time parameter. The Reverberance parameter controls the density and smearing of both the early reflections and the reverberation cloud.

High frequency energy tends to fade away first as a sound is dissipated in a room. The High Frequency Decay Factor adjusts the time it takes for the high frequency energy to die away and thus changes the characteristics of the room. Rooms with smooth, hard surfaces are more reflective and have less high frequency damping. Rooms filled with sound absorbing materials, such as curtains or people, have more high frequency damping.

The Low Frequency Decay Factor parameter adjusts the time it takes for the low frequencies to die away. This control adjusts the "boominess" of the room.

Parameter	Description
Decay Time	Sets the reverb decay time. Range: 0% to 100%
HF Decay Factor	Sets the rate at which high frequencies die away. The high frequencies last longer as the percentage is increased. Range: 0% to 100%
LF Decay Factor	Sets the rate at which low frequencies die away. The low frequencies last longer as the percentage is increased. Range: 0% to 100%

Parameter	Description
Early Reflections	Sets the volume of the initial wall reflections. Range: 0% to 100%
Reverberance	Sets the amount of scattering of the early reflections and the reverberation cloud. Range: 0% to 100%

Mono Delays - 100, 250, 500, 750, 1500, 3000

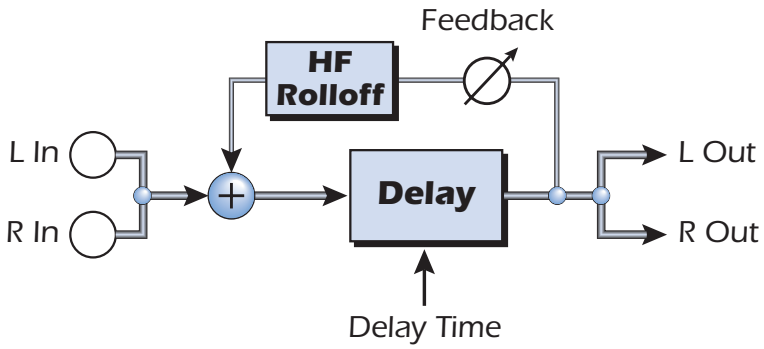
A delay line makes a copy of the incoming audio, holds it in memory, then plays it back after a predetermined time. The delay number refers to the maximum delay time that can be produced by the delay line. The six lengths, from 100 ms to 3 seconds, allow you to make the most efficient use of the effect memory resource.

Long delays produce echoes, short delays can be used for doubling or slapback effects. Very short delays can be used to produce resonant flanging and comb filter effects or create monotone robotic-sounding effects (Hint: use feedback). Stereo signals are summed together before entering the Mono Delay.

There is also a feedback path to send the delayed audio back through the delay line. When creating echo effects, the feedback controls how many echoes will be produced. With short delays, the feedback control acts as a *resonance* control, increasing the amount of comb filtering produced by the delay line. Comb filtering: See page 57.

A High Frequency Rolloff filter in the feedback path cuts some of the high frequency energy each time the audio goes through the delay line. This simulates the natural absorption of high frequencies in a room and can also be used to simulate tape-based echo units.

The Wet/Dry mix controls how loud the echoes are in relation to the original signal.



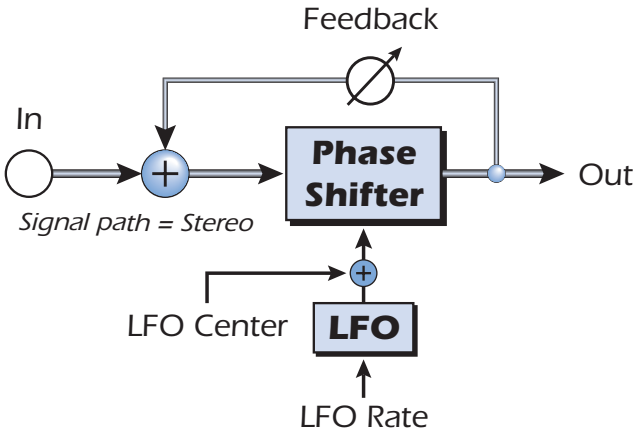
Parameter	Description
Delay Time Mono Delay 100 Mono Delay 250 Mono Delay 500 Mono Delay 750 Mono Delay 1500 Mono Delay 3000	Sets the length of the delay in milliseconds. *(.01ms. minimum increment between settings)* Range: 1 millisecond to 100 milliseconds Range: 1 millisecond to 250 milliseconds Range: 1 millisecond to 500 milliseconds Range: 1 millisecond to 750 milliseconds Range: 1 millisecond to 1.5 seconds Range: 1 millisecond to 3 seconds
Feedback	Sets the amount of delayed signal that will be recirculated through the delay line. Range: 0% to 100%

Parameter	Description
High Freq. Rolloff	Damps high frequencies in the feedback path. Range: 0% to 100%

Phase Shifter

A phase shifter produces a fixed number of peaks and notches in the audio spectrum which can be swept up and down in frequency with a low frequency oscillator (LFO). This creates a swirly, ethereal sound with harmonically rich sound sources of a type of pitch shift with simpler sounds. The phase shifter was invented in the 1970's and the characteristic sound of this device evokes emotions of that musical era.

By setting the LFO Depth to zero and tuning the LFO Center, a fixed multi-notch filter is created.



Parameter	Description
LFO Center	Sets the initial offset of the LFO and changes the position of the peaks and notches. Range: 0% to 100%
Feedback	Increases the depth of the notches and height of the peaks. Range: 0% to 100%
LFO Rate	Controls the sweep rate of the Low Frequency Oscillator. Range: .01Hz to 10Hz
LFO Depth	Controls how much the Center Frequency is swept by the LFO. Range: 0% to 100%
Waveform	Selects a Sine or Triangle wave for the LFO
LFO L/R Phase	Controls the stereo width by adjusting the phase difference between the left and right sweeps. Range: -180° to +180°

Rotary

This is a simulation of a rotating speaker used on organs. The rotating speaker was invented to give static organ tones a pipe organ type of animation, but this distinctive sound became a legend in its own right. Spinning a sound around the room creates a doppler pitch shift along with many other complex and musically pleasing sonic effects.

The Rotary incorporates acceleration and deceleration as you switch between the two speeds.

Parameter	Description
Speed	Switches between slow or fast rotor speeds with acceleration and deceleration as the speed changes.

Speaker Simulator

The Speaker Simulator provides realistic guitar speaker responses and is designed for use with guitar, bass or synthesizer. Twelve popular guitar amp speaker cabinets are modeled.

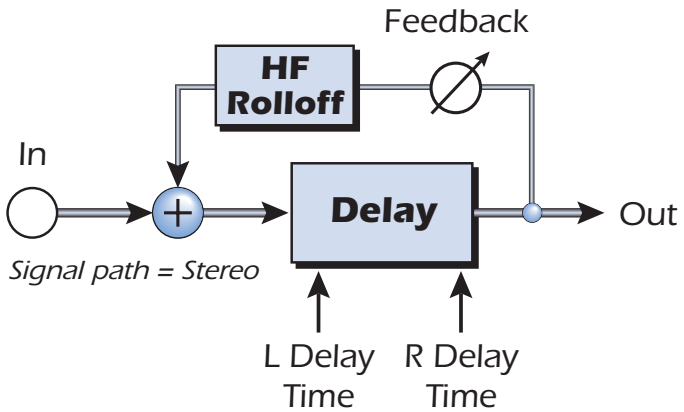
There is only one parameter on this effect. Just select the speaker you want and listen. Normally this effect should be used with the Mix control set to 100%.

Speaker Type	Description
British Stack 1 & 2	Modeled from a British 8-speaker high power amplifier stack.
British Combo 1-3	Modeled from a British 2-speaker combo amplifier.
Tweed Combo 1-3	Modeled from an American, 1950's era, 2-speaker combo amplifier.
2 x 12 Combo	Modeled from an American, 1960's era, 2-speaker combo amplifier.
4 x 12 Combo	Modeled from an American, 1960's era, 4-speaker amplifier set.
Metal Stack 1 & 2	Modeled from a modern era, power amplifier stack.

Stereo Delays - 100, 250, 550, 750, 1500

The Stereo Delays are true stereo delay lines in that the left and right channels are kept entirely separate from each other. The delay number refers to the maximum delay time that can be produced by the delay lines. The five different lengths, from 100 ms to 1.5 seconds, allow you to make the most efficient use of the effect memory resource.

Because the left and right channels can have different delay times, you can create a panning effect by setting one delay long and the other short. Very short delay times combined with a high feedback amount can be used to create monotone robotic-sounding effects. Using the longer stereo delays, you can "overdub" musical lines one on top of the other with the feedback control turned up.

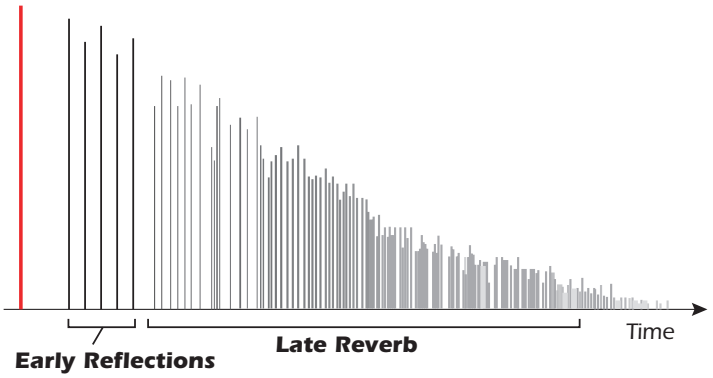


Parameter	Description
Left Delay Time	Sets the length of the delay for the left channel in milliseconds.
Right Delay Time	Sets the length of the delay for the right channel in milliseconds.
Delay Time (L & R) Stereo Delay 100 Stereo Delay 250 Stereo Delay 550 Stereo Delay 750 Stereo Delay 1500	*(.01ms. minimum increment between settings)* Range: 1 millisecond to 100 milliseconds Range: 1 millisecond to 250 milliseconds Range: 1 millisecond to 550 milliseconds Range: 1 millisecond to 750 milliseconds Range: 1 millisecond to 1.5 seconds
Feedback	Sets the amount of delayed signal that will be recirculated through the delay line. Range: 0% to 100%
High Freq. Rolloff	Damps high frequencies in the feedback path. Range: 0% to 100%

Stereo Reverb

Reverberation is a simulation of a natural space such as a room or hall. The stereo reverb algorithm is designed to simulate various halls, rooms and reverberation plates.

Decay time defines the time it takes for the reflected sound from the room to decay or die away. The diagram below shows a generalized reverberation envelope.



After a short pre-delay period, the echoes from the closest walls or ceiling are heard. These first echoes, or early reflections, vary greatly depending on the type of room. Some time after the early reflection cluster ends (late reverb delay), the late reverberation (a dense cloud of complex wall reflections) begins and decays according to the time set by the Decay Time parameter.

Diffusion is the amount of scattering and density of the late reverberation cloud. Rooms with many complex surfaces have more diffusion than bare rooms.

High frequency energy tends to fade away first as a sound is dissipated in a room. The High Frequency Damping parameter adjusts the time it takes for the high frequency energy to die away and thus changes the characteristics of the room. Rooms with smooth, hard surfaces are more reflective and have less high frequency damping. Rooms filled with sound absorbing materials, such as curtains or people, have more high frequency damping.

The Low Frequency Damping parameter adjusts the time it takes for the low frequencies to die away. This control adjusts the “boominess” of the room.

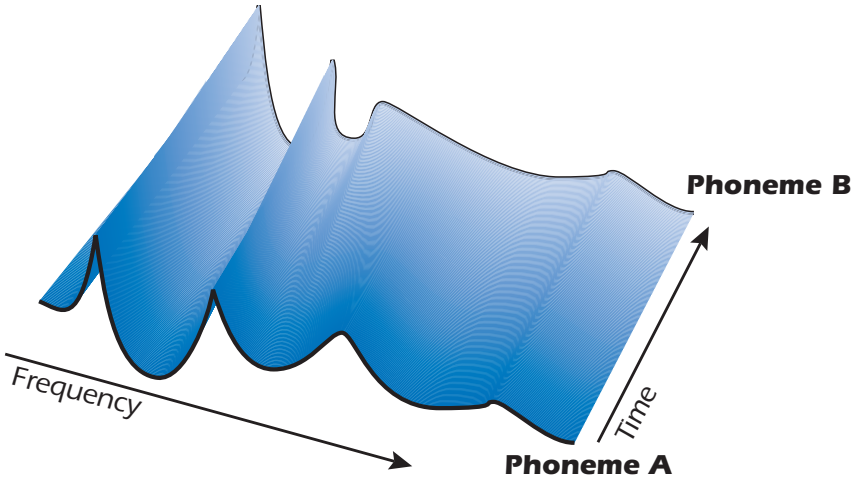
Parameter	Description
Decay Time	Sets the length of the Late Reverb. Range 1.5 to 30seconds
Early Reflections Level	Sets the volume of the initial wall reflections. Range: 0% to 100%
Early/Late Reverb Bal	Adjusts the balance between early refections and late reverb. Range: 0% to 100%
Late Reverb Delay	Sets the time between early reflections and the onset of the late reverb cloud. Range: 1ms to 350ms
Diffusion	Sets the amount of scattering of the late reverb cloud. Range: 0% to 100%
High Freq. Damping	Sets the rate at which high frequencies die away. Range: -10.0 to +3.0 damping factor
Low Freq. Damping	Sets the rate at which low frequencies die away. Range: -10.0 to +3.0 damping factor

Vocal Morpher

This unique effect allows you to select two vocal phonemes and morph between them using an LFO. Phonemes are the consonants and vowels we use in articulating speech sounds and these sounds are very distinctive and evocative. 30 different phonemes are available and these can be shifted up or down in pitch for even more effects.

To use the Vocal Morpher, you just select Phoneme A and Phoneme B from the list of thirty. Now the LFO automatically morphs back and forth between the two selected phonemes, creating interesting vocal articulations. The rate of the LFO is adjustable and you can select between Sine, Triangle or Sawtooth waveforms. The sine and triangle waves fade smoothly. The sawtooth wave gradually fades, then jumps abruptly back.

When the frequency of the A or B Phonemes is shifted up or down, entirely new effects can be produced. These frequency controls can also be used to tune the phoneme frequencies to the range of audio you are processing.



List of Available Phonemes

A	**E**	**I**	**O**	**U**	**AA**
AE	**AH**	**AO**	**EH**	**ER**	**IH**
IY	**UH**	**UW**	**B**	**D**	**F**
G	**J**	**K**	**L**	**M**	**N**
P	**R**	**S**	**T**	**V**	**Z**

Parameter	Description
Phoneme A	Select any of the available Phonemes for Phoneme A.
Phoneme A Tuning	Adjusts the frequency of Phoneme A up or down 2 octaves in semitone intervals. Range: -24 semitones to +24 semitones
Phoneme B	Select any of the available Phonemes for Phoneme B.
Phoneme B Tuning	Adjusts the frequency of Phoneme B up or down 2 octaves in semitone intervals. Range: -24 semitones to +24 semitones
LFO Rate	Controls how fast the phonemes morph back and forth. Range: .01Hz to 10Hz
LFO Waveform	Selects the waveform for the morph: Sinusoid, Triangle, Sawtooth

Gate

This stereo noise gate is useful both for background noise reduction applications and also for special effects.

The gate uses an envelope follower and threshold detector to turn on its output when the input signal is above the turn-on threshold, and shut down its output when the signal falls below the shut-off threshold. When “turned on” the Gate passes the input signal through to the output at unity gain and when “shut off” the Gate silences the output or attenuates it by an adjustable gain factor. While the Gate is a stereo effect, the left and right signals are gated in unison, with the envelope follower defaulting to the louder of the two signals.

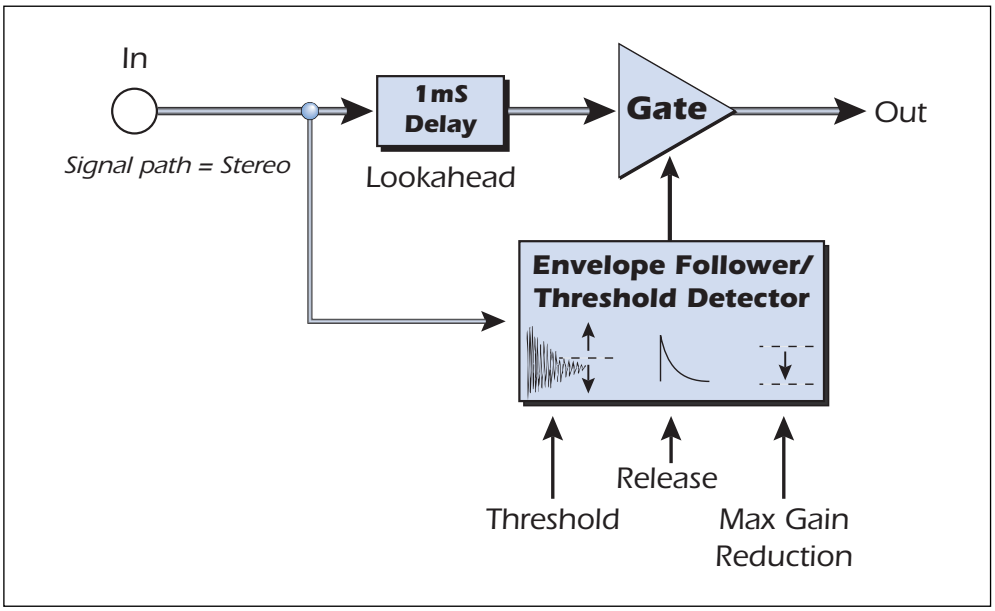
In normal operating mode, Gate turn-on is nearly instantaneous when the input signal exceeds the Threshold level, while Gate Release time is an adjustable parameter. The effect of the fast turn-on can be enhanced by using an optional 1 millisecond lookahead in the Gate's envelope detector.

Together with the Threshold setting, tuning the Release time parameter is very useful in order to achieve the least-obtrusive, most natural-sounding gating effect, which is highly dependent on the specific program material being processed.

The gate does not offer an adjustable wet/dry mix parameter but does supply a Bypass switch for effectively removing the effect from the signal path.

Applications

- Basic Gating - reduce background noise during periods of low signal level
- Re-Enveloping - extreme release time/attenuation can be used to re-sculpt the signal envelope
- Punch Enhancement - high threshold+fast shuttoff+modest attenuation perform an expander-like function that accentuates transients



The Gate behaves exactly as a straight wire, except when activated by a signal level below the Threshold (with Lookahead Off).

Parameters

Threshold

When the input signal rises above the level set by the Threshold parameter, the Gate is triggered to turn on and go from its maximum gain reduction level up to 0dB gain. The turn-on threshold is adjustable anywhere between -70dB and 0dB (below the PatchMix nominal operating point of -12dBFS.)

One of the keys to the smooth operation of the Gate is that the input Threshold level that turns on the Gate is always higher than the level that shuts off the gate. **This means that the input signal level must descend substantially below the Threshold in order to turn off again.**

This difference between turn-on and shut-off levels, or the hysteresis, is 10dB. That means that if the Threshold is -30dB, the signal level must fall to -40dB before the Gate will begin to shut off.

Release Time

This parameter controls the time, in milliseconds, that is required for the Gate to shut off. More specifically, this is the time that will be required for the Gate control signal to go from unity gain at 0dB down to the Max Gain Reduction level.

The optimum value for the Release time is dependent on the program material as well as the effect you're trying to achieve. Optimum Release time is also highly dependent upon the settings of the Threshold and Max Gain Reduction parameters.

In general, times less than about 10 msec are prone to cause clicks in the output, while times longer than 30 msec may make the gating effect obvious if the background signal being gated out is very noisy.

Max Gain Reduction

This parameter sets the attenuation that will be applied to the signal when the Gate is shut off. The Gate control signal will swing between 0dB and this value as the Gate turns on and shuts off.

To perform a strict “gating” operation, Max Gain Reduction would normally be set to -infinity in order to completely silence the output of the Gate.

However, there are good reasons to set Max Gain Reduction to something less drastic than infinite attenuation. Sometimes the silence between gated signals is “too quiet” - especially when the signal represents a solo vocal or instrument, where the complete lack of any sound between voiced segments sounds unnatural. For these applications, setting Max Gain Reduction somewhere between -20dB and -40dB is more appropriate.

In tandem with a high Threshold, Max Gain Reduction can also be set to very modest values like -5 or -10dB in order to add a subtle “punch” enhancement to transients. This has an effect similar to an expander, where the attack transients which exceed the Threshold stand out by 5 or 10dB above the normal signal (you can make up for that 5 or 10dB attenuation by using a trim pot or boosting the channel strip gain after the Gate.)

Lookahead

By default, the Gate effect uses a fixed 1 millisecond lookahead to avoid clipping off the leading edge of signal transients when the Gate turns on. However, this is actually implemented by adding a 1 millisecond delay to the signal through the gate. For applications where this additional 1 millisecond latency is a problem, the Lookahead can be turned Off.

Level Meter

This meter represents the input signal level in dB, and is in fact the output of the Gate's envelope follower. Since the envelope follower is driven by the greater of the left or right channel, this monophonic meter represents the greater of the two input signals.

Gain Reduction Meter

This meter shows the value in dB of the gate control signal which is used to boost or attenuate the input signal. Its most-rightward maximum value of 0dB represents a unity gain path through the Gate in its turn-on state. Except for the possibility of the 1 millisecond lookahead latency, the Gate behaves exactly as a straight wire in this turned-on state. Values less than 0dB represent the amount by which the input signal is being attenuated as the Gate shuts off.

The most-leftward gain shutoff value achieved by the Gain meter is set by the Max Gain Reduction parameter (values from -70dB to -infinity are off the meter.) The speed with which the Gain signal decays from 0dB to the shutoff value can be observed to change according to the Release time parameter.

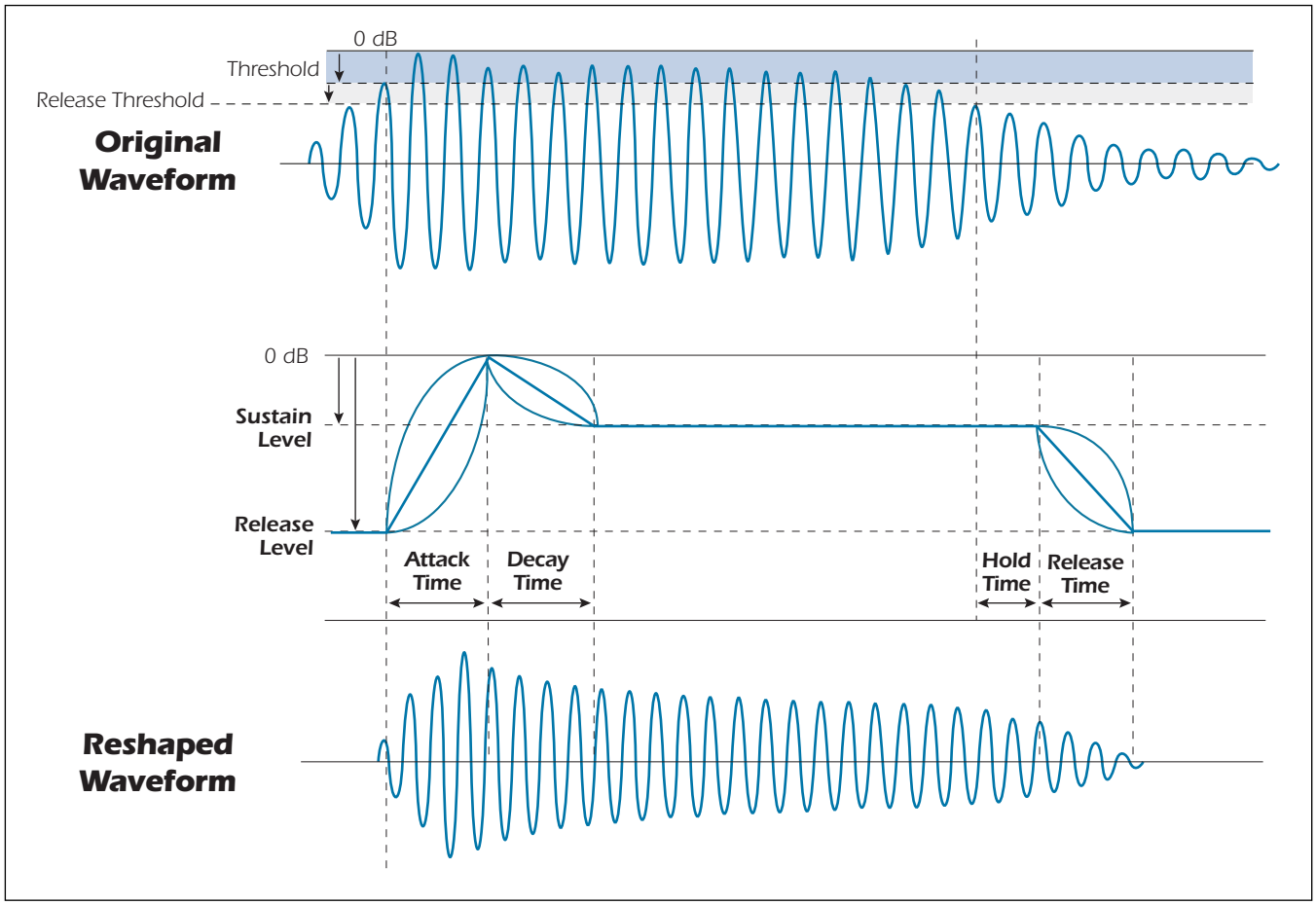
Reshaper

The Reshaper effect is a special purpose dynamics modification program, designed to "resculpt" the amplitude envelope of an audio signal. The effect uses an envelope follower and threshold detector to drive an ADSR-type gain stage, which can impose new attack, decay, sustain and release profiles on the signal's original envelope.

Applications

- "Punch" Reducer - slow turn-on with added lookahead trims attacks off signals
- "Punch" Enhancement - fast turn on with high thresholds and release gain expands signal attack transients
- Auto Volume Pedal - long attack times with Attack Retrigger can automatically simulate use of a guitar volume pedal for gently fading in each note.
- Ambience Reduction - can be used like a gate to suppress ambient reverberations that below a certain threshold.

When the input signal exceeds an adjustable Threshold, the Attack phase begins and continues until the gain reaches unity (0dB). After the Attack peaks, the gain stage immediately transitions into the Decay phase, which continues until the gain falls to the Sustain level. During the Sustain phase, the gain stage holds a constant level until the input signal passes below the Release Threshold. During the Release Phase, the gain returns to the Release Level where it remains until the another input transient triggers the next Attack phase.



Attack, Decay and Release times are all adjustable, and the shape of each of these segments is selectable between exponential, linear, or logarithmic. An additional Hold Time can be used to extend the Sustain phase past the point where the signal has passed the Release Threshold.

If the Sustain Level is set the same as the Release Level, then the Reshaper effectively becomes a two-phase “transient catcher” where the Release Threshold, Hold Time and Release Time are ignored.

While the peak Attack gain level is always fully turned on, note that the Release Level is not necessarily completely off, but can be adjusted upward so that the Reshaper retains a nominal minimum gain. This allows the Reshaper to resculpt only the louder transients of a signal while maintain a nominal output signal level the rest of the time.

The Release Threshold is always expressed relative to the Attack Threshold so that they will automatically track each other when the Attack Threshold is adjusted.

Parameter	Description
Attack Threshold	When the input signal rises above the level set by the Attack Threshold parameter, Reshaper's ADSR engine begins the Attack phase. The turn-on threshold is adjustable anywhere between -40dB and 0dB (below the PatchMix nominal operating point of -12dBFS.)

Parameter	Description
Attack Time	This parameter controls the time, in milliseconds, that is required during the Attack phase for the gain rise from its quiescent Release Level to unity gain, or 0dB.
Decay Time	This parameter controls the time, in milliseconds, that is required for the gain to fall from 0dB down to the attenuated Sustain Level. Note that if the Sustain Level is set to 0dB this decay time becomes simply a delay before entering the Sustain phase.
Release Time	This parameter controls the time, in milliseconds, that is required for the gain to fall from the Sustain Level down to the Release Level.
Level Meter	This meter represents the input signal level in dB, and is in fact the output of the Gate's envelope follower. Since the envelope follower is driven by the greater of the left or right channel, this monophonic meter represents the greater of the two input signals.
Sustain Level	This sets the gain level applied to the input signal when the ADSR engine is in the Sustain phase.
Release Level	This sets the final gain level applied to the input signal when the Release phase is fully decayed. When set to the minimum (-70dB) the effective Release Level is -infinity, i.e., fully turned-off.
Hold Time	This parameter allows additional time to be added onto the Sustain phase after the input signal falls below the Release Threshold before transitioning to the Release phase. This extension of the Sustain phase is useful for altering the tail dynamics of the sound.
Attack Lookahead/ Delay	This parameter is adjustable in milliseconds to allow the Reshaper to either "look ahead" and advance (negative values) or "delay" (positive values) the response of the envelope detector relative to the dynamics of the input signal. For example, negative lookahead values can cause the envelope detector to start the ADSR's Attack phase before the actual attack of the signal so as not to miss any audible transients. Likewise, positive delay values can be used to start the Attack "late", so that signal transients are intentionally missed by the Attack.
Release Threshold	This parameter controls the level in dB below the Attack Threshold at which the Release phase of the ADSR will begin.
Attack Retrigger	By default, when the value of this parameter is Disabled, the Reshaper's ADSR engine will wait until at least the Release phase of a cycle before restarting a new Attack phase. By setting Attack Retrigger to Enabled, however, the Reshaper becomes sensitive to new input signal transients during any phase of the ADSR cycle. In addition, enabling this parameter will also cause the attack to restart at the Release Level instead of whatever gain was being applied when the new attack arrived.
Attack Curve	This parameter allows the gain during the Attack phase to follow one of 3 curves: linear, logarithmic, or exponential. Because the ADSR computes gain using linear coefficients, the exponential curve comes the closest to being a "constant in dB" gain ramp. A linear curve provides a somewhat more immediate turn-on, while the logarithmic curve presents a very abrupt turn-on.

Parameter	Description
Release Curve	This parameter selects gain curves exactly as for the Attack Curve parameter, except that the selected curves apply to both the Decay and Release phases of the ADSR.

Multimode EQ

The Multimode EQ is a flexible stereo filter that is capable of implementing a range of powerful filter topologies. It is useful both for utility EQ applications and also for special effects.

The Multimode EQ is built from an array of filter sections that can be configured to support:

- Lowpass filters with up to 48dB/octave rolloff
- Highpass filters with up to 48dB/octave rolloff
- Highpass + Lowpass series or parallel combination with up to 24dB/octave rolloff
- Bandpass filters with up to 24dB/octave rolloff
- Bandcut filters with up to 24dB/octave rolloff

In addition to cutoff or center frequency parameters, each of the above filter types also has a switchable rolloff rate and adjustable resonance.

A Filter Edit parameter controls whether the Multimode EQ operates in Stereo, where filter parameters are adjusted identically for both channels, or split Left and Right, where the left and right channels support completely independent filter types and parameter values.

In addition to a standard Bypass switch, the effect offers an adjustable wet/dry mix parameter. While not normally found on EQ sections, adjustable wet/dry mixtures can be useful for generating phase cancellation and other special effects.

Applications

- Basic Tone Control - for fidelity enhancement
- Rumble Filter - use the highpass configuration with 48dB/octave rolloff below 50Hz.
- Subwoofer Support - use the lowpass configuration with 48dB/octave rolloff below 100Hz.
- Lo-Fi Effect (telephone, walkie-talkie, guitar mini-amp, distance simulation)
- Extreme Spectral Shaping - use Highpass+Lowpass, Bandpass or Bandcut with independent hi/lo resonance
- Pseudo-stereo Effect - apply slightly different EQ to left and right channels to broaden the spread of a mono signal
- Cross-over - left and right channels split a mono signal between highpass and lowpass with a sharp transition region.

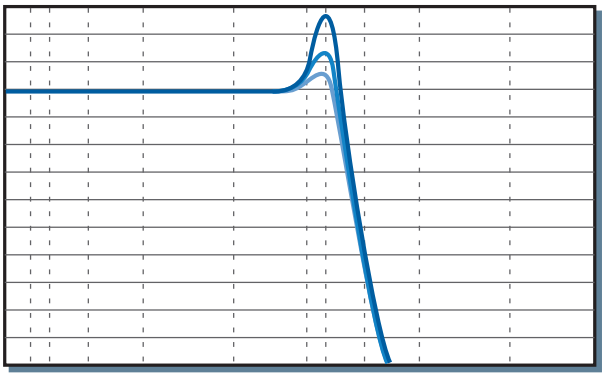
Parameters

While the Multimode EQ has many parameters applicable to the various possible configurations of channels and filters, it selectively enables or hides parameters depending on their applicability to the current configuration. As a result, not all of the parameters listed below appear on-screen at the same time.

Parameter	Description
Filter Edit	This parameter controls whether the filter editing parameters apply to both left and right channels in tandem (Stereo), only to the left channel (Left) or only to the right channel (Right).
Filter Mode	This parameter selects one of 5 different filter types: Lowpass, Highpass, Lowpass+Highpass, Band Pass or Band Cut.

Lowpass

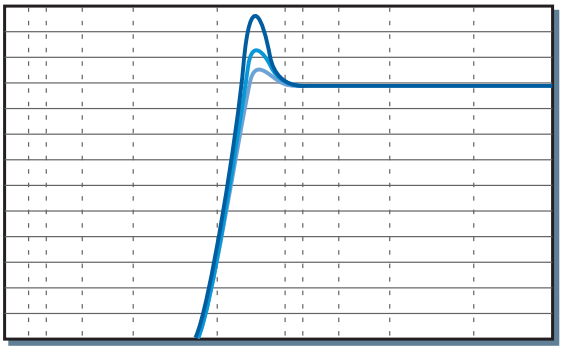
The frequency response of the lowpass filter looks something like the diagram below:



In this mode, the Lowpass filter can have up to a 48dB/octave rolloff slope. In this mode the Lowpass Rolloff, Lowpass Frequency and Lowpass Resonance parameters are available for editing the filter response.

Highpass

The frequency response of the highpass filter looks something like the diagram below:

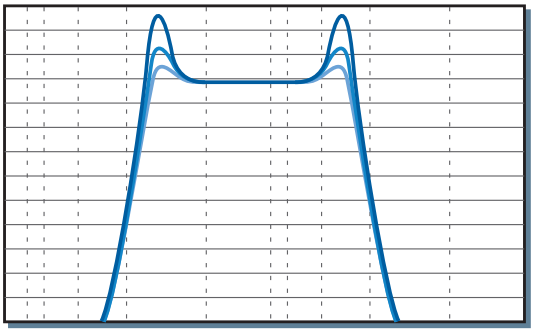


In this mode, the Highpass filter can have up to a 48dB/octave rolloff slope. The Highpass Rolloff, Highpass Frequency and Highpass Resonance parameters are available for editing the filter response.

Highpass -> Lowpass

In this mode, the Lowpass and Highpass filters are connected in series and both sets of Lowpass and Highpass parameters are exposed and independently editable to create the overall filter response. The maximum rolloff slope of each filter is limited to 24dB/octave in this mode.

In Highpass -> Lowpass mode, the effect does not place any limitations on the Frequency parameters of one filter relative to the other. In normal use, the Highpass Freq parameter will be less than the Lowpass Freq parameter, creating a bandpass-type response:



However, if the Highpass Freq parameter is greater than the Lowpass Frequency parameter, the passband effectively disappears, since the part of the spectrum which is above the highpass and below the lowpass is non-existent. As a result, you'll hear a rapidly attenuating bandpass response as the corner frequencies diverge.

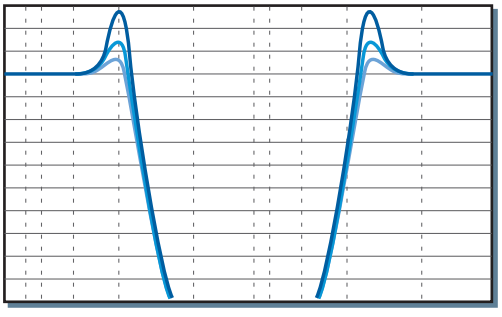
Note that while the Highpass -> Lowpass combination appears the same as the Band Pass filter, this mode is different in several important respects:

- The rolloff points are independently adjustable as individual frequencies rather than specified as a combination of center frequency and bandwidth.
- The rolloff slope of each High and Low filter can be specified separately while the Bandpass and Band Cut filters use the same slope.
- The Resonance of each High and Low filter can be specified separately while the Bandpass filter uses the same Resonance at high and low corner frequencies.

Highpass || Lowpass

In this mode, the Lowpass and Highpass filters are connected either in parallel, and both sets of Lowpass and Highpass parameters are exposed and independently editable to create the overall filter response. The maximum rolloff slope of each filter is limited to 24dB/octave in this mode.

In Highpass || Lowpass mode, the effect does not place any limitations on the Freq parameters of one filter relative to the other. In normal use, the Highpass Freq parameter will be higher than the Lowpass Freq parameter, creating a bandcut-type response:



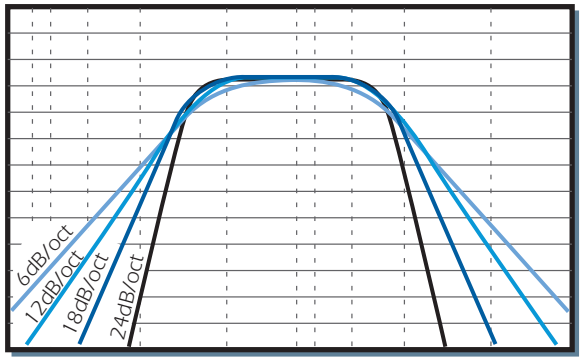
However, when the Highpass Freq parameter is lower than the Lowpass Freq parameter, the combined filter response is basically flat, since the passbands of each filter combine to admit the entire spectrum. An exception occurs when there is resonance added to the filters - you'll hear the resonant peaks as increased gain above the otherwise flat spectral response.

Note that while the Highpass || Lowpass combination appears the same as the Band Cut filter, this mode is different in several important respects:

- The rolloff points are independently adjustable as individual frequencies rather than specified as a combination of center frequency and bandwidth.
- The rolloff slope of each High and Low filter can be specified separately while each side of the Band Cut filter uses the same slope.
- The Resonance of each High and Low filter can be specified separately while the Band Cut filter uses the same Resonance at high and low corner frequencies.

Band Pass

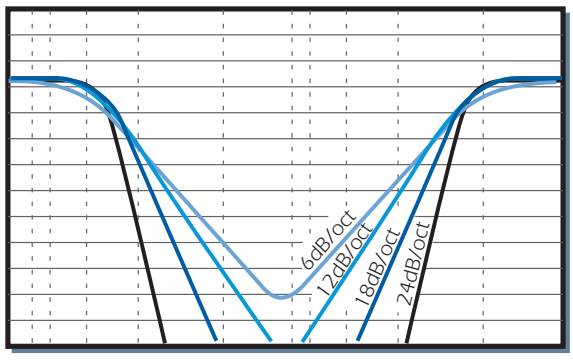
In this mode, the Lowpass and Highpass filters are connected in series to form a bandpass filter, whose Center Freq and Bandwidth parameters are used to generate the rolloff frequencies for the underlying Lowpass and Highpass filters. In this mode the rolloff slope on the high and low sides of the passband is symmetrical and is limited to a maximum of 24dB/octave. The Resonance is also common to both filter sections .



Resonance = 0

Band Cut

In this mode, the Lowpass and Highpass filters are connected in parallel to form a band-cut filter, whose Center Freq and Bandwidth parameters are used to generate the rolloff frequencies for the underlying Lowpass and Highpass filters. In this mode the rolloff slope on the high and low sides of the cut-band is symmetrical and is limited to a maximum of 24dB/octave.The Resonance is also common to both filter sections .



Resonance = 0

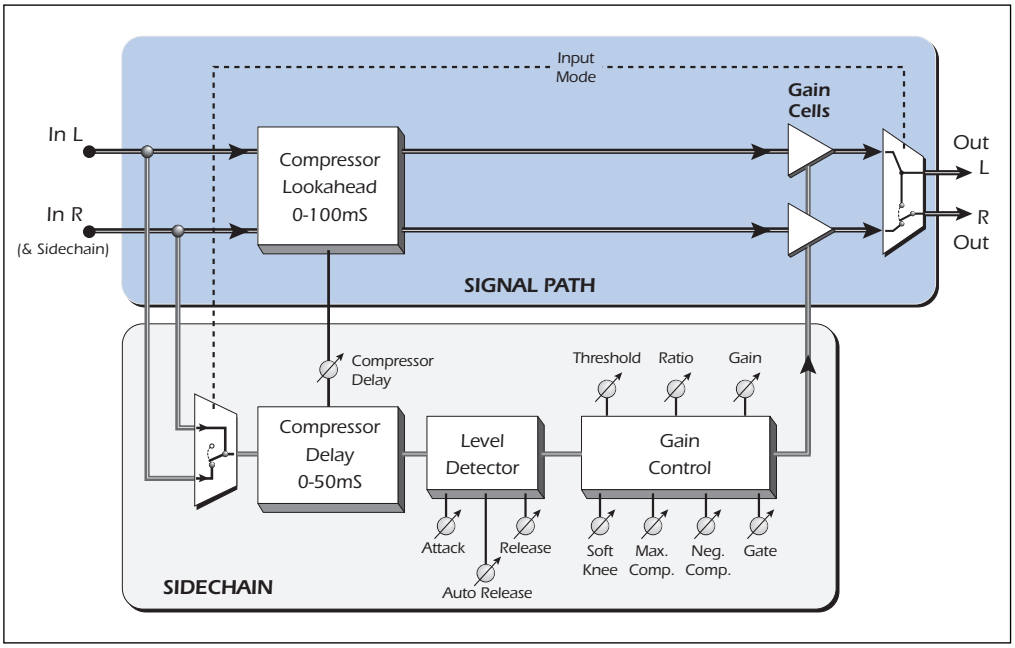
RFX Compressor

The RFX Compressor is a full-featured stereo compressor effect which features the standard parameters available on most compressors as well as a collection of additional advanced parameters that are useful for more sophisticated applications and special effects:

- Threshold, Ratio, Attack and Release w/gain metering
- Auto-makeup gain
- Adjustable soft knee
- Adjustable lookahead/delay
- Noise gate (downward expander)
- Compressor “tail” expansion
- Program-dependent release
- Negative compression ratios

Signal Flow

The block diagram of the RFX Compressor is shown below.



Note that the effect is split between a signal path and a sidechain path that contains the compressor's level detectors and gain computation. The signal path of the RFX Compressor is very close to a “straight wire”, with only a delay line and one gain control element inserted in it. The sidechain contains the bulk of the compressor algorithm and is responsible for computing the gain control signal. Signal multiplexers at the front of the signal path and sidechain allow linked stereo compression or split signal path/sidechain processing.

The RFX Compressor does not have the input gain control that is found on some compressors. These are typically used to align the input signal range to the compression threshold. Instead, we've allowed the RFX Compressor's Threshold parameter to operate over an exceptionally large range of 0-60dB so that it can be “steered” to the appropriate range of the input signal. The output Gain parameter also operates - either manually or automatically - over the

unusually large range of -60dB to +60dB in order to renormalize the compressor's output for the next stage of the signal path.

The wide dynamic range of the RFX Compressor aside, it's generally a good idea to maintain the hottest signal levels possible without clipping at the input to any audio processor.

Parameters

Threshold

Threshold sets the input signal level above which dynamic range compression takes place. Everything above the threshold will be brought down in volume. The compression threshold ranges from -0 to -60dB, relative to full scale (0dBFS) input.

Setting the Threshold to 0dB disables normal compression, since no signal can exceed the maximum possible input level. A Threshold setting of 0dB is still useful, however, when using soft-knee compression or gating, since these actions occur below (and their thresholds are set relative to) the Threshold parameter.

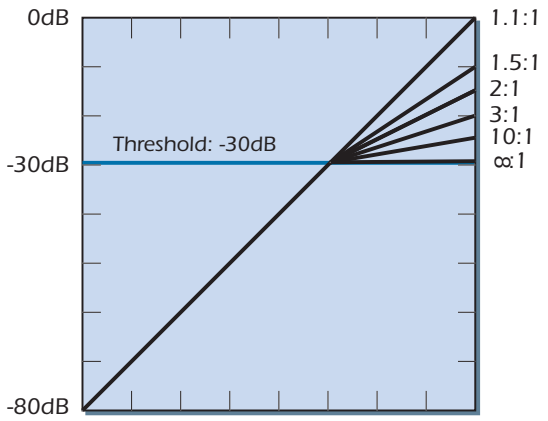
Gain Reduction Meter

As input signals exceed the Threshold, the rightness character in the bargraph is lit, and successive characters are lit for each approximately 3dB in gain reduction imposed by the compressor on the input signal. Because this is a compression meter and not a level meter, the same input signal level will show widely varying meter readings depending on the setting of the Ratio parameter.

Ratio

Sets the ratio of output signal to input signal levels, selectable in 16 steps from 1:1.1 to 1:INFINITY.

When Neg Compression is set to Enabled, the range of compression ratios extends beyond INFINITY to encompass negative compression ratios from 1:-100 down to 1:-1, which can be useful for applications like ducking and other special effects.
See the discussion of the Neg Compression parameter on page 82.



● **Tip:** *A ratio of Infinity:1 combined with high threshold and fast attack/release results in an effective peak limiter.*

Attack

Sets the amount of time that the compressor's level detector will take to respond to an increase in signal level. The Attack range is adjustable from Instantaneous (essentially a peak detector that follows individual samples) to 10 seconds (useful for long-term leveling or automatic mixing applications.)

Release

Sets the amount of time that the compressor's level detector will take to respond to a decrease in signal level. The fastest Release time is 100 microseconds, useful for some special effects but highly prone to distortion; more typical release times are in the range of 70 milliseconds to 1 second. Release times up to 10 seconds are available for long-term leveling or automatic mixing applications.

When the Auto-release parameter is in its signal-dependent settings, the Release time shown represents the shortest possible release time. In Auto-release modes the displayed Release time will be automatically extended depending on the dynamics of the input signal.

Gain

Sets the compressor's output gain in dB, from +60dB boost to-60dB cut. This control follows all of the other elements in the compressor's signal path, so positive gain boost can be used to make up for the gain reduction normally applied to signals above the compression threshold. Alternatively, negative gain cut can be used to make up for the gain increase that is applied to signals below the threshold in Soft Knee mode.

▼ ***Caution!*** *The Gain control can increase the signal level to the point of clipping. Excessive signal levels can damage speakers as well as your ears!*

Auto Makeup Mode: When adjusted downward past the -60dB cut, the Gain parameter begins operating in Auto Makeup mode. Auto Makeup mode is used to compensate for the drop in output level normally resulting from the gain reduction actions of the Threshold and Ratio parameters. Auto Makeup makes it much easier to adjust these parameters since there is no need to switch back and forth to the Gain parameter in order to perform the gain compensation manually.

Auto Makeup looks at the gain reduction implied by the setting of the Threshold and Ratio parameters and automatically applies a complementary gain increase so that an ideal 0dB input signal results in a 0dB - or lower - output signal. In this mode, indicated by the Threshold legend, Thresh -54dB the Gain parameter adjusts the output level from that 0dB input signal to fall anywhere in the range of 0dB down to -60dB.

▼ *Auto Makeup should not be used when in negative compression ranges (see the Neg Compression parameter on page 82. Use manual Gain control instead.*

Advanced Parameters

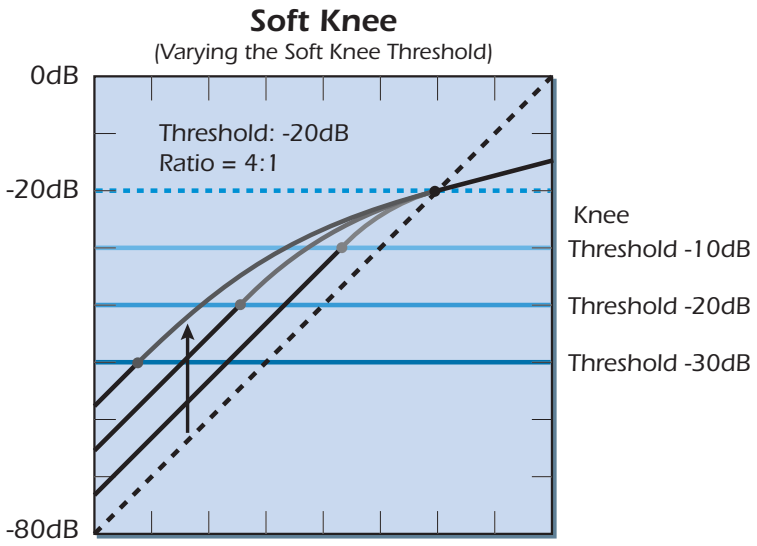
This parameter controls whether the “Advanced Parameters” listed in this section are hidden or exposed on the screen. For simple applications, quick edits or for novice users, these advanced parameters can be hidden to minimize screen clutter and preclude erroneous operation. For special and exotic applications and for experienced users, these parameters can be exposed to allow access to all the gory details of the compressor's operation.

Note that even when this parameter is set to “Off”, the settings of the advanced parameters are still active; the only effect of this parameter is to hide them from the screen.

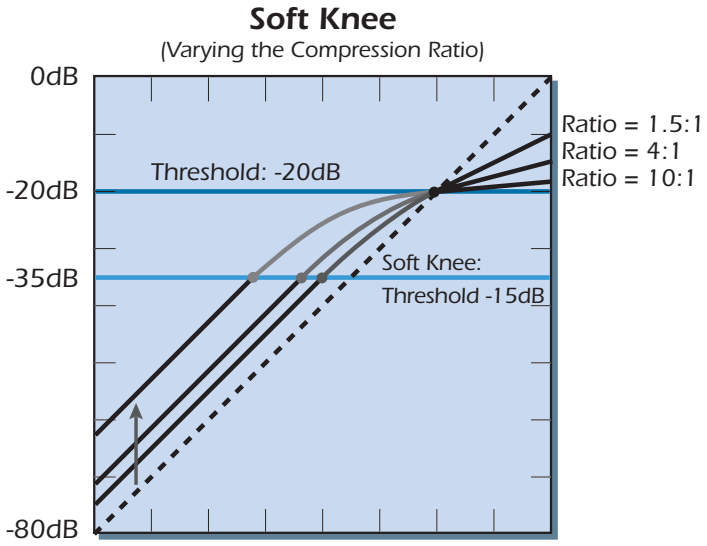
Soft Knee

This parameter sets the depth of the compression transition region, giving an adjustable hard or soft “knee” to the compressor's gain curve. Setting the depth of this region results in a knee shape that can be varied from a sharp transition to one that is imperceptibly gradual.

With the default value of Off, the Soft Knee parameter causes the gain curve to switch immediately at the Threshold point from no compression (1:1) to full compression (1:Ratio), representing the hard knee effect. By adjusting the parameter value, an additional knee threshold can be created 1dB to 60dB below the regular compression Threshold. Between these two thresholds the effective compression ratio increases smoothly along the curve of a circular arc, from 1:1 at the lower knee threshold to the full compression of 1:Ratio at the upper Threshold. Both the Soft Knee depth and the Ratio will affect the particular shape of the knee: shallower depths and higher Ratios will create a sharper knee, while greater depths and lower ratios create a softer knee.



This diagram shows the effect of varying the Soft Knee Threshold. *Compression is 1:1 (no compression) at the Knee Threshold and smoothly transforms into the selected compression ratio at the Compression Threshold. The upward arrow shows the additional gain added to signals below the Threshold.*



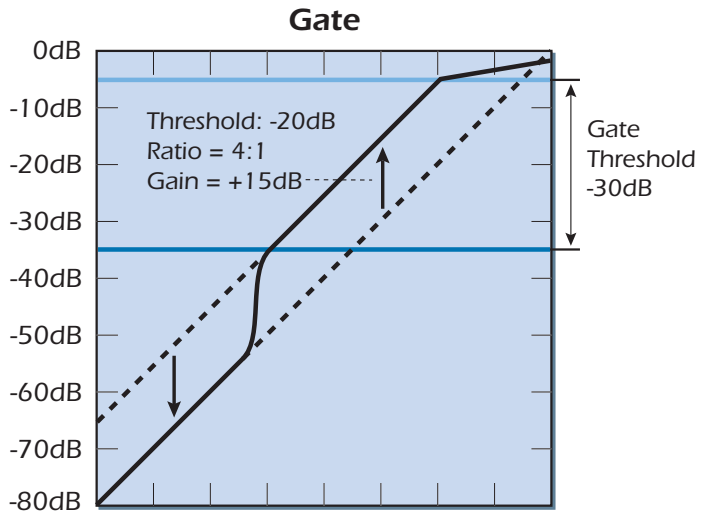
This diagram shows the effect of varying the Compression Ratio *with a fixed Soft Knee Threshold. The knee transforms from a linear slope to the slope of the compression ratio over the Soft Knee Threshold area. The upward arrow shows the additional gain added to signals below the Threshold.*

In the region between the lower knee and upper Threshold, a variable amount of gain reduction is applied depending on the signal level and Ratio setting. To keep this gain reduction from "dragging down" the signal levels at the Threshold point, a complementary gain boost is automatically applied to all signal levels below the Threshold when the Soft Knee is enabled. This gain increase with depth and Ratio is illustrated by the upward arrows in the diagrams, and is similar to the action of the Auto Makeup Gain parameter. Thus signal levels below the Threshold increase as the Soft Knee depth and/or Ratio is increased (but see the Gate parameter, below.)

● **Tip:** *Setting a high Ratio with the Threshold at 0dB and the Soft Knee at -60dB creates a compressor whose ratio varies smoothly from gentle compression at lower signal levels to peak limiting at maximum signal level*

Gate

This parameter enables automatic gain reduction on signals that fall from 1 to 120dB below the Threshold point (or Soft Knee threshold, if enabled.) This can act effectively as a “noise gate” on low-level signals that have been boosted by the action of the Gain or Soft Knee parameters. The gating action follows a somewhat soft-kneed contour of its own so that turn-on and turn-off at the gate threshold is not too abrupt.

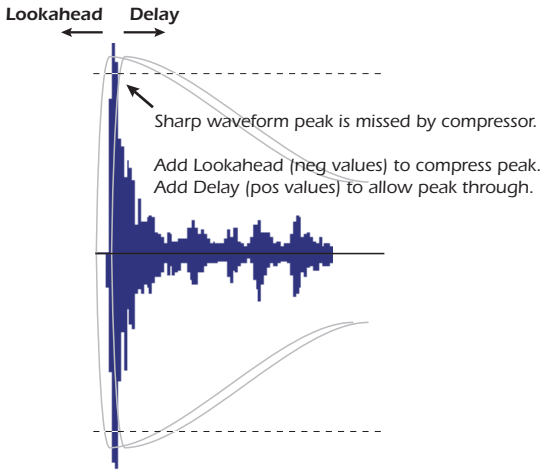


In this example, the Gain has been boosted by +15dB. The Gate cancels out the +15dB Gain boost below the Gate Threshold. Signal levels above the Gate Threshold will be boosted; signal levels below this point will not be boosted and will be 15dB lower in volume.

Note that, strictly speaking, the term “gate” is a misnomer in this context, since the action of this parameter is simply to cancel out gain increases that resulted from the settings of other parameters. This can be seen by the arrows in the diagram as the gain is reduced below the Gate threshold back down to the dotted line representing unity gain. The result is that if the Gain parameter is set negative or the Soft Knee parameter is disabled, the Gate parameter will have no effect.

Comp Lookahead/Delay

This parameter controls compressor lookahead or delay by setting the relative time offset, in milliseconds, between the compressor's signal path and its sidechain path.


Lookahead
Delay
Sharp waveform peak is missed by compressor.
Add Lookahead (neg values) to compress peak.
Add Delay (pos values) to allow peak through.

At negative values, this parameter lets the level detector in the compressor's sidechain “look into the future” up to 100 milliseconds in order to anticipate upcoming peaks in the signal - accomplished of course, by inserting delay into the signal path. This lookahead technique allows the use of slower attack times without missing signal peaks.

At positive values, the signal path delay is zero; instead, a delay of up to 50 milliseconds is inserted into the sidechain path containing the level detector. This delay can be used intentionally to cause the compressor to miss signal

peaks, retaining the “punch” and “bite” of signal attacks while subsequently compressing the sustained portions of the sound.

In general, both positive and negative values of this parameter are useful for applications where the normal envelope of a signal is being creatively manipulated to achieve special effects.

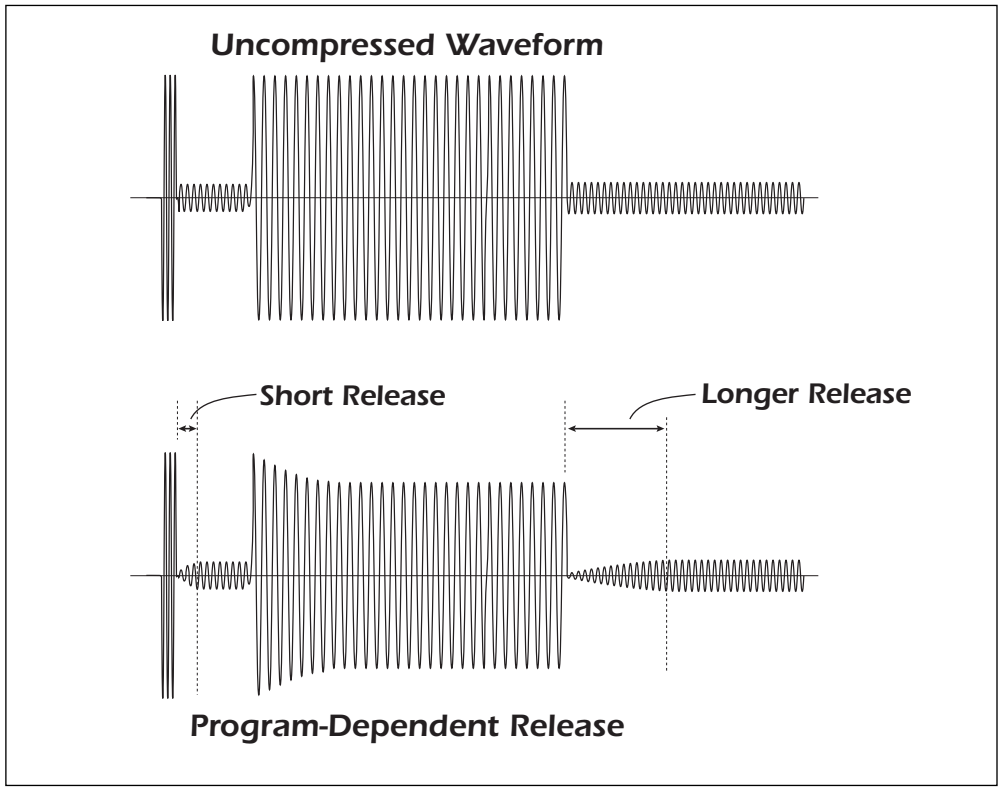
Auto-Release

This parameter causes the effective Release time to be extended automatically based on the dynamics of the input signal. This parameter emulates the program-dependent release characteristics found on some classic analog compressor/limiters.

When not set “Off”, the Auto-release parameter treats the Release parameter value as a minimum release time, extending it by as much as a factor of 10 depending on different, selectable characteristics of the input signal:

In Program-dependent mode, release times are increased depending on how often, how long and by how much the input signal (“program material”) exceeds the Threshold. Release times increase slowly under sustained excursions of the input over the Threshold, and typically return back to normal within a few seconds after the signal level has fallen below it. This emulates the signal “memory effect” exhibited by some electro-optical compressors.

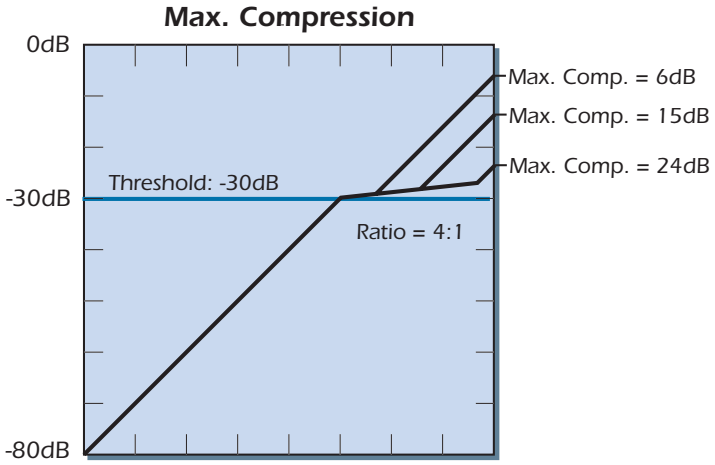
In Compression-dependent mode, the release extension characteristics are similar, but in addition depend on the amount of gain reduction being applied to the signal. Thus the same signal will cause more release-time extension at higher compression Ratio settings than at lower ones.



With Auto-release turned on, the release time becomes longer after an extended period of compression.

Max Compression

This parameter is used to limit the amount of gain reduction that the compressor can apply. The limit is set as a maximum number of dB of gain reduction, from 3dB to UNLIMITED.



This feature emulates the phenomenon of the compression "tail" found in the gain curves of some classic analog compressor/limiters. The phenomenon results from the inability of these devices to apply more than a certain amount of compression to the input signal. When the device "runs out" of enough gain reduction to compress a very high level signal, it resumes a 1:1 gain curve again. This "deficiency" has the unexpected sonic benefit of restoring some dynamics to the compressed signal - but only on the highest input peaks - thus adding some "life" back into otherwise over-compressed signals.

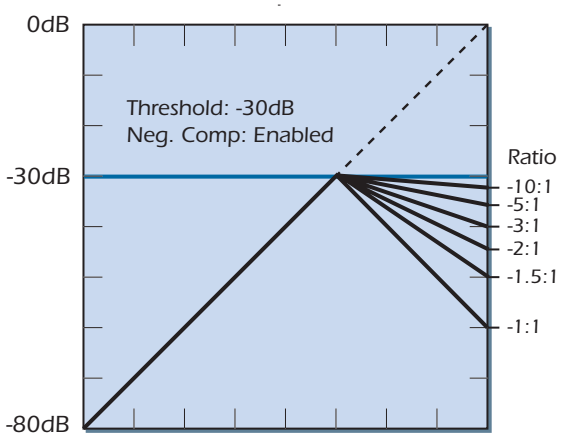
Unlike analog compressors, the Max Compression parameter allows you to adjust the amount of gain reduction before the compressor returns to a 1:1 gain curve. The diagram shows three settings of the Max Compression parameter; the compressor "gives up" and returns to 1:1 after 6, 15 and 24dB of compression have been exhausted, respectively.

The parameter is most useful at higher compression ratios, allowing the gain curve to be carefully tailored to the dynamics of the signal as well as the Threshold and Ratio parameters. The limit set by the Max Compression parameter does not apply to gain reduction performed in the Soft Knee region of the gain curve.

■ **Note:** *You may need to use the Gain parameter to keep these restored peaks from clipping the compressor output since Auto Makeup gain doesn't automatically take the compressor tail into account.*

Neg Compression

When the Neg Compression parameter is **Enabled**, the range of compression values available to the Ratio parameter extends beyond INFINITE to encompass negative compression ratios from 1:-100 down to 1:-1. Using negative compression ratios results in an output signal that actually gets quieter as the input signal rises above the threshold. This action can be useful for applications like ducking and for other special effects.



The diagram above shows the gain curves using a Threshold at -30dB and a range of negative compression ratios.

At just past 1:INFINITE, the setting of 1:-100 causes input signals approaching 0dB to be only slightly decreased below -30dB. In contrast, the compression ratio of 1:-1 causes a 2dB gain reduction for each 1dB of additional input signal level, resulting in an output signal level that is folded down over the Threshold.

Input Mode

The Input Mode parameter allows the compressor signal path and sidechain to be driven in common or by separate inputs. This is a feature of many compressors and is useful for a range of applications and special effects.

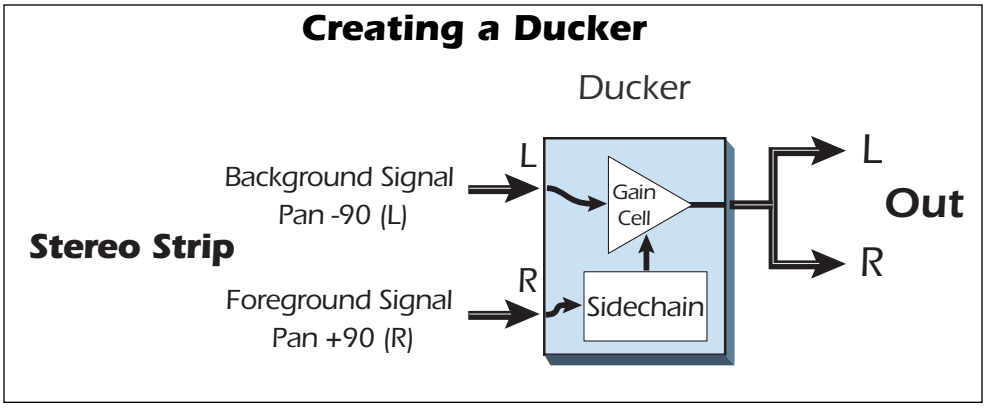
By default, the Input Mode of the compressor is **Stereo**. In this mode the two independent left and right signal paths are gain controlled by a parallel sidechain path common to both inputs that contains the compressor's level detector. This single level detector works on the higher of the two input signal levels, so that signal peaks are properly compressed and no L/R image shift results from compression operations.

When the Input Mode is set to **L In/R Sidechain**, the signal path is fed exclusively from the left channel and the sidechain is fed exclusively from the right channel. This allows dynamics control between two completely independent signals. In this mode both the compressor's left and right outputs are fed by the mono signal from the left input channel's signal path.

Splitting the signal path and sidechain makes possible applications where the two signals may be completely unrelated, such as ducking. Other split-sidechain applications result from situations where a stereo input signal has had different processing applied between left and right channels. One example would be to place a stereo equalizer ahead of the compressor in order to implement a version of de-essing or “de-booming”. See page 86.

▶ Create a Ducker

To create a *ducker*, in which a background signal's level is reduced in the presence of a foreground signal, first set the Input Mode parameter to L In/R Sidechain. Then send feeds from the background signal to the left input, and from the foreground signal to the right side input. Set the Ratio parameter to -1:1 (or lower for less background reduction), and dial in a low Threshold such as -50, so any foreground signal above -50dB will cause gain reduction in the background signal. This technique works best with slow Attack and Release times — use a liberal amount of Compression Lookahead to keep the background from masking the beginning of foreground sounds.



Example Settings

Here we have provided a few examples to show the varied uses of this useful tool. Bear in mind that these examples are simply starting points and that you will undoubtedly need to fine tune the parameters to fit the program material and to suit your own taste.

▶ Increase Drum Punch:

Adjust the Threshold control to control the amount of compression.

- **Threshold:** Adjust so that all hits are being compressed.
- **Ratio:** 4:1
- **Attack:** 8 msec (Increase the time to hear more "stick" sound.)
- **Release:** 60 msec (Adjust according to the tempo of song.)
- **Gain:** Adjust to make up for lost volume.
- **Soft Knee:** Adjust as desired.
- **Comp. Lookahead:** This can be used instead of the Attack control.
- **Max. Compression:** Unlimited

▶ Smoothing out the Bass Guitar Level:

This setup evens out the volume and prevents the bass guitar from wandering in and out of the mix.

- **Threshold:** -24dB (adjust according to the sound)
- **Ratio:** 4:1
- **Attack:** 8 msec
- **Release:** 70 msec
- **Gain:** +4dB (adjust according to the sound)
- **Soft Knee:** Threshold -8dB
- **Gate:** Off
- **Comp. Lookahead:** 0 msec
- **Auto-release:** Comp-dependent
- **Max. Compression:** 18dB

▶ Peak Limiting:

This setup trims only the very loudest peaks, leaving most of the signal intact.

- **Threshold:** -37dB (adjust according to the sound)
- **Ratio:** 2:1 or 3:1
- **Attack:** Instantaneous
- **Release:** 30 msec
- **Gain:** 0dB
- **Soft Knee:** Off
- **Gate:** Off
- **Comp. Lookahead:** -5 msec
- **Max. Compression:** Unlimited

▶ *Vocal Compression/Spoken Word:*

This setup compresses the entire dynamic range of the vocal. Whenever there is a signal present, there is some compression taking place.

- ***Threshold:*** Adjust so that the first bar of the meter comes on even on soft passages.
- ***Ratio:*** 2:1
- ***Attack:*** 0.1 msec
- ***Release:*** 100 msec
- ***Gain:*** Set to compensate for lost gain.
- ***Soft Knee:*** Off
- ***Gate:*** Off
- ***Comp. Lookahead:*** 0 msec
- ***Auto-release:*** Off
- ***Max. Compression:*** 12dB

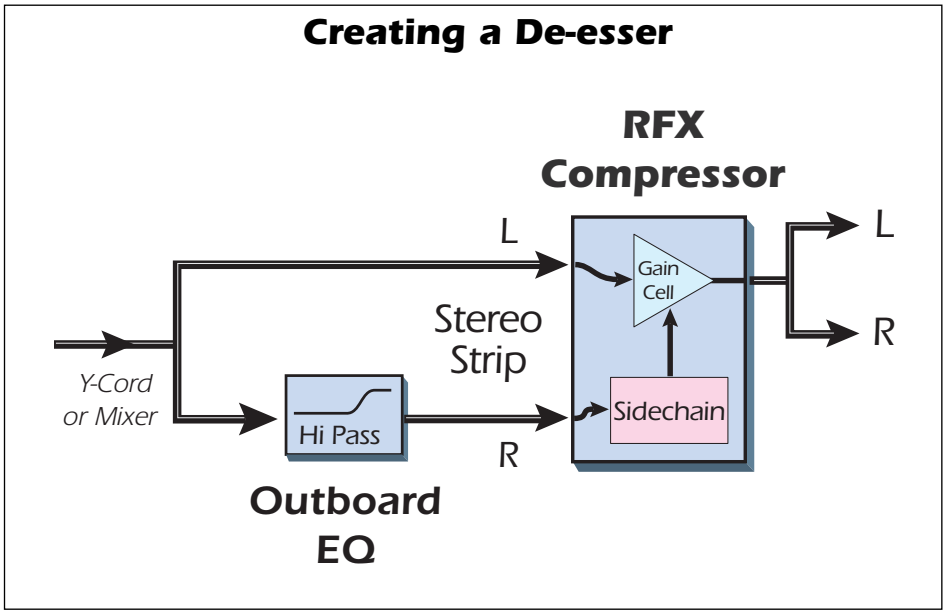
▶ *Backwards Drums & Cymbals:*

This is a special effect which reverses the volume envelope of cymbals and drums.

- ***Threshold:*** -37dB (adjust according to the sound)
- ***Ratio:*** -1:1 (Neg. Compression enabled)
- ***Attack:*** Instantaneous
- ***Release:*** 200 msec
- ***Gain:*** +19dB
- ***Soft Knee:*** Off
- ***Gate:*** Off
- ***Comp. Lookahead:*** -24 msec
- ***Auto-release:*** Off
- ***Max. Compression:*** Unlimited

▶ Creating a De-esser:

A de-esser reduces the sibilance or "sss" sound in a vocal part. The sidechain feature of the RFX Compressor makes it possible to create a effective de-esser using the compressor and an external highpass filter or graphic EQ. The idea is to boost the high frequency content going to the sidechain (R input) so that the compressor will turn down the volume in the presence of sibilance.



Connections and EQ Settings

1. Connect an outboard highpass filter or equalizer to the Right Input.
2. Connect your preamplified vocal signal to the outboard EQ *and* the Left Input.

■ You can boost the low frequencies in the right channel to create a "de-boomer".

Compressor Settings

Lookahead gives the compressor time to react to the vocal sibilance.

- **Input Mode:** L In/R Sidechain
- **Threshold:** -32dB (adjust to control amount of de-essing)
- **Ratio:** 2.5:1
- **Attack:** Instantaneous
- **Release:** 40 msec
- **Gain:** 0dB
- **Soft Knee:** Off
- **Gate:** Off
- **Comp. Lookahead:** -20 msec
- **Auto-release:** Off
- **Max. Compression:** Unlimited

E-MU PowerFX

The hardware-accelerated effects of t
w you to use PatchMix DSP effects from within Cubase, Sonar, or other host application with minimal load on your CPU.

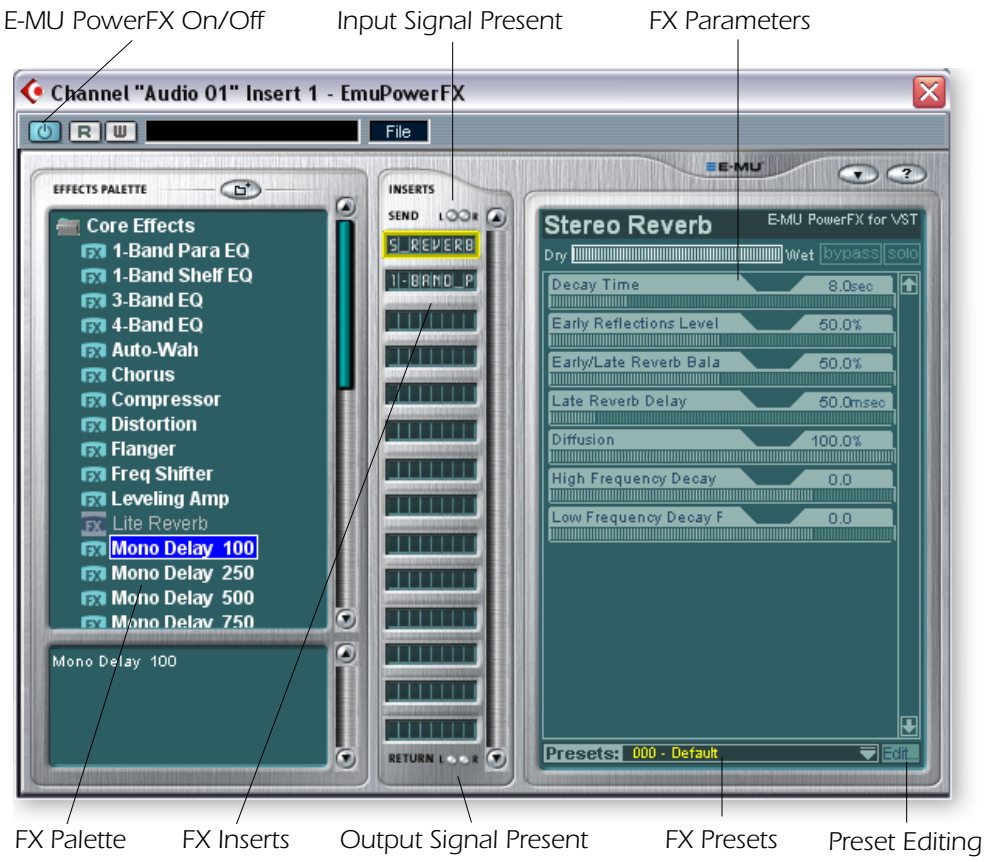
Note: PowerFX is not supported in Vista. While many users are abble to use E-MU PowerFX under Vista with little or no problems, we are unable to offer support to those using PowerFX with Windows Vista.

E-MU PowerFX incorporate smart time alignment technology which automatically compensates for system latencies and ensures proper synchronization of audio throughout the VST chain (if the host application supports this feature).

▼ E-MU PowerFX are not available at the 96kHz or 192kHz sample rates.

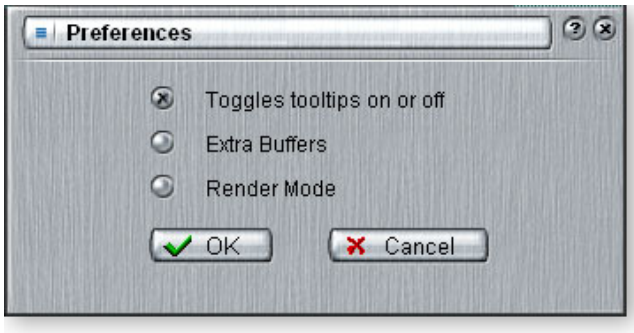
● Cubase SX/SL/LE 2.0, Nuendo and Sonar (using the Cakewalk VST adapter 4.4.1) implement VST 2.X auto delay compensation.

● When using Cubase LE with the 0404, turn Multiprocessing OFF in Cubase LE (Device Setup, VST Multitrack, Expert).



Parameter	Description
PowerFX On/Off	Enables or bypasses E-MU PowerFX.
FX Palette	Select from a single “Core” effect or a Multi-Effect.
FX Inserts	Drop Effects from the FX Palette here.
Signal Present LEDs	These indicators turn blue to show the presence of input and output signals.
FX Parameters	Select the desired effect in the center insert section, then adjust the wet/dry mix and parameters for the effect.
FX Presets	Select from the list of preprogrammed effect presets here.

Parameter	Description
Preset Editing	Click here to Save, Delete, Rename or Overwrite a User Preset. See the "User Preset Section" for more information
Preferences	The Preferences menu allows you to: ▪ Toggle the Tooltips On or Off ▪ Extra Buffers - Check this box if excessive stuttering occurs when using E-MU PowerFX in your VST Host application. This box should be checked when using Fruity Loops. ▪ Render Mode - Induces realtime rendering in applications which do not support realtime rendering. (WaveLab, SoundForge)



▶ To Setup & Use E-MU PowerFX:

▼ *Using any driver other than "E-MU ASIO" may produce undesirable results when using E-MU PowerFX.*

Setup Cubase LE

1. Launch Cubase LE
2. Instantiate **E-MU PowerFX** in an Insert or Aux Send location within Cubase.
3. Press the Insert Edit button in Cubase to bring up the E-MU PowerFX plug-in window shown on the previous page.

Setup E-MU PowerFX

4. Make sure the **Insert Enable** button is illuminated, indicating that E-MU Power FX is on. The blue "Signal Present" indicators will be illuminated if E-MU PowerFX is properly patched into a signal path.
5. Drag the desired effects from the Effects Palette to the center Insert strip.
6. Click on the Effect you wish to edit in the center Insert Strip *(it will highlight in yellow)*, then adjust the effects parameters in the right section of the window.
7. You can also select or edit User Presets from the section below the FX parameters. See the "User Preset Section" for more information.

Add Delay Compensation *(if needed)*

If you are using Cubase VST 5.1, or another older sequencer without automatic delay compensation, you will have to insert an E-Delay Compensator into any other audio tracks to keep them time-aligned.

8. Simply insert an E-Delay Compensator plug-in into the same insert location you used for E-MU PowerFX on any other audio tracks. That's it.

Automating E-MU PowerFX

E-MU PowerFX can be automated in Cubase LE (or other recording host) just like any other VST effect. When "Write Automation" is activated in Cubase, control changes made in the PowerFX window during playback will be recorded on a special "Automation Subtrack". When "Automation Read" is activated, the recorded control changes will be played back.

▼ Steinberg Cubasis does not have the control automation feature.

▶ To Record E-MU PowerFX parameter changes in Cubase LE

1. Add E-MU PowerFX as a Channel Insert.
2. Rewind the song and enable "Automation Write" by pressing the WRITE button on, illuminating it. (Refers to Cubase LE. If you are using another application, refer to the documentation.)
3. Bring the E-MU PowerFX window to the front and select the Effect you want to automate. The effect parameters appear in the TV screen. Make sure the blue "On" button is lit.
4. Press the **Play** button on the Cubase Transport control. The song begins playing.
5. Adjust the E-MU PowerFX controls to achieve the effect you want. Rewind the song when finished.
6. Disable "Automation Write" and enable "Automation Read" . Playback the song to hear and view your changes.
7. To edit Automation, first enable both "Automation Write" and "Automation Read" and press **Play**. Cubase LE begins overwriting as soon as you change a control.
8. If you don't like the results and want to try again, select **Show Used Automation** from the Project menu. The Automation Subtrack appears. Next, click in the Parameter Display and select **Remove Parameter**.

   Note: This only erases one automation parameter from the Automation Subtrack. To erase multiple control edits, repeat the procedure above. See the Cubase LE manual for more specific information about automation editing.

▼ Once you have recorded or drawn automation, do not delete or move effects from the Insert Strip. Doing so will result in unpredictable behavior.

E-MU PowerFX Resource Availability

Because different collections of VST plug-ins and PatchMix Sessions can be run simultaneously, it is possible to load a new plug-in for which resources are not available. If DSP resources are NOT available for an existing setup:

- E-MU PowerFX loads a Hardware I/O Path and simply passes audio through without any effects. The effects insert slot(s) in PowerFX will be "redded out".
- If no Hardware I/O Paths are available, the plug-in will be disabled and run in a soft pass-through mode. The effects insert slot(s) in PowerFX will be "grayed out".
- If DSP resources ARE available, but no Hardware I/O Paths are available, the plug-in will run in soft pass-through mode.
- If the sample rate is changed in the middle of a PowerFX session, E-MU PowerFX plug-ins will be bypassed, since the hardware effects cannot operate at 96kHz or 192kHz.

E-MU PowerFX Compatibility Chart

Application Name	Compatible?	Note	Render	Extra Buffers
Steinberg Cubase VST 5.1	Yes		Off	Off
Steinberg Cubase SX 1	Yes		Off	Off
Steinberg Cubase SX 2	Yes	Instrument Freeze triggers error if not in render mode.	Off	Off
Steinberg Cubase LE	Yes		Off	Off
Steinberg Cubase SL	Yes		Off	Off
Steinberg WaveLab 4	Yes		On	Off
Steinberg WaveLab Lite (ver 4)	Yes		On	Off
Steinberg WaveLab 5	No	Pops & clicks may occur. (Try 8 buffers at 1024)	On	Either
Sony Acid 4	Yes		On	Off
Sony Vegas 5	Yes		On	Off
Sony SoundForge 7	No	Power FX crashes when launched.	On	Off
Adobe Audition 1.5	No	Audio distortion & immediate lockup.	Any	Any
FruityLoops Studio 4.5	Yes		Off	On
Ableton Live 3.5	No	Distortion when FX parameters are changed.	On	Off
Cakewalk Sonar 3	Yes		Off	Off

Rendering Audio with E-MU PowerFX

Rendering (sometimes called Export) is a mixdown process performed by the host application, which creates a new digital audio file from a multitrack song. Rendering allows a virtually unlimited number of VST effects to be used because the audio processing is performed out of realtime.

E-MU PowerFX and the PatchMix DSP effects are strictly realtime processes. When E-MU PowerFX are used while rendering audio, the rendering process must proceed at realtime rate. Some host applications are not designed to handle realtime rendering and this can cause problems. E-MU PowerFX can be used with these applications if you are willing to follow certain guidelines.

General Tips for Rendering using E-MU PowerFX

- If an error message occurs, increase the "ASIO Buffer Latency" setting located in the device Setup dialog box. Depending on your setup, you may have to increase or decrease the Buffer Latency settings to find the setting that works.
- Instead of rendering with E-MU PowerFX, bounce the E-MU PowerFX processed tracks to another track in realtime.
- Check "Realtime Render" in the Render dialog box when using Cubase LE, Cubase SX2 or Cubase SL2. This setting will give the best results.

Tips for using Freeze Mode on Cubase LE

- Make the project length as short as possible. Freeze always renders the entire project length, even if the MIDI track being rendered is shorter.
- **Great Tip:** Temporarily bypass E-MU PowerFX (and any other effects) even when "Freezing" another track. This will allow the track to Freeze faster than realtime.

Using E-MU PowerFX with WaveLab and SoundForge

Stuttering in the audio can occur when rendering with SoundForge or any version of Steinberg WaveLab. This problem is caused by discontinuities in the first few audio buffers as they are fed by WaveLab to E-MU PowerFX. The problem can be eliminated by following these guidelines.

- Check "Render Mode" box in the E-MU PowerFX preferences. See page 88.
- We recommend that you only use the MME/WAVE E-DSP Wave [xxxx] drivers.
- Reduce the "Buffer Size" in the WaveLab, Audio Preferences dialog box. This moves the stuttering to beginning of the file.
- Pad the beginning (and/or end) of your audio file with silence (.5 to several seconds depending on the file). This action causes the buffer discontinuities to occur *before* the song begins.

E-MU VST E-Wire

E-Wire is a special VST/ASIO Bridge which allows you to route digital audio via ASIO to PatchMix and back again.

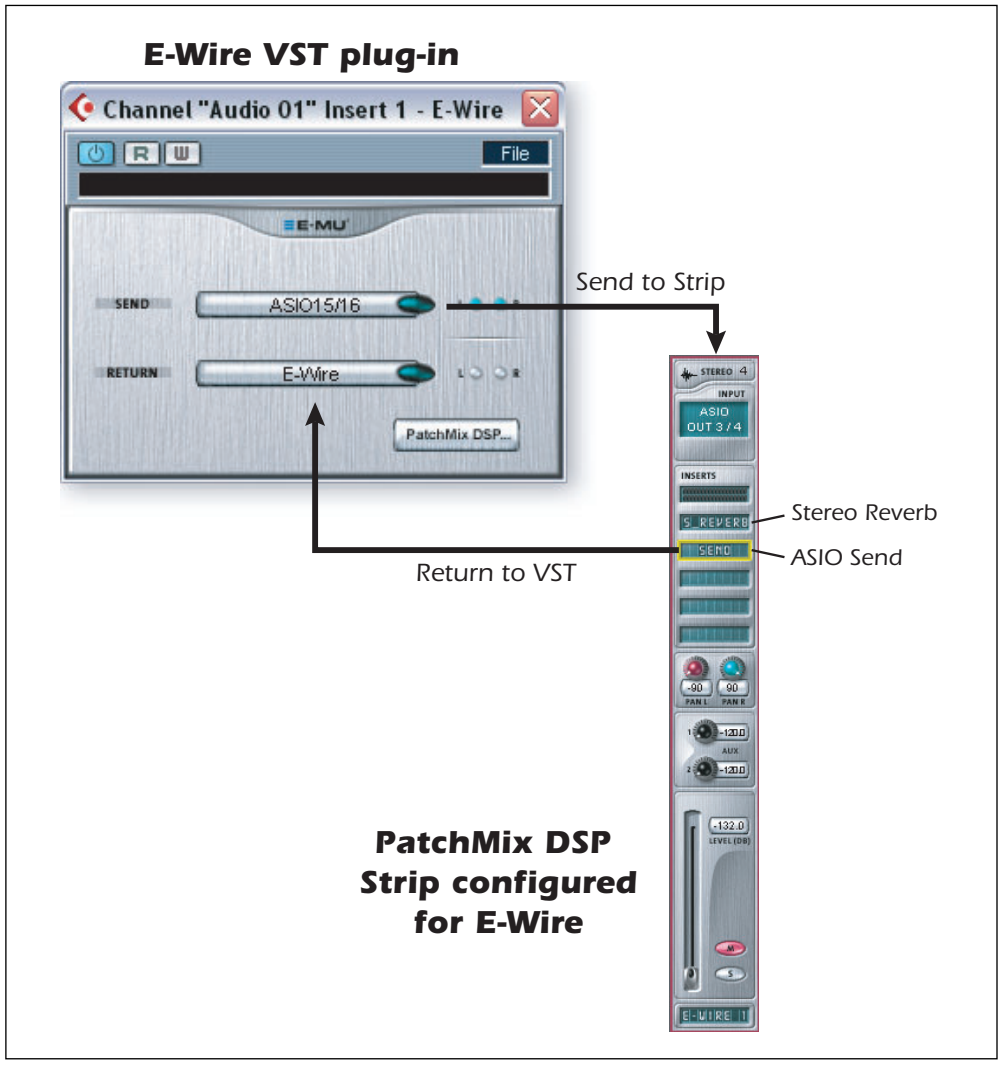
E-Wire VST incorporates smart time alignment technology that automatically compensates for system latencies and ensures proper synchronization of audio throughout the VST chain. In addition, E-Wire also allows you to insert outboard audio gear into the VST environment.

E-Wire has three main components:

- A VST plug-in which handles the audio routing to PatchMix DSP.
- An ASIO mixer strip in PatchMix DSP configured to route audio to the E-Wire plug-in. You simply drop the effects you want to use into this strip.
- For hosts that don't support automatic delay compensation, a manual delay-compensation plug-in can be inserted in Cubase tracks or channels that don't use E-Wire to compensate for ASIO delay.

The diagram below may give you a better idea of how E-Wire works:

■ **Note:** *It's easier to use E-MU PowerFX instead of E-Wire if you just want to use the hardware effects. (E-Wire was the precursor to E-MU PowerFX.) However, E-Wire can be very useful because it allows you to route VST inserts or Sends to Physical Inputs and Outputs via PatchMix DSP.*



E-Wire bridges the gap between hardware I/O and the VST world. The E-Wire VST plug-in sends audio to a strip containing the desired effect. An ASIO Send routes the audio back to E-Wire VST.

▶ To Setup and use E-Wire:

Setup PatchMix DSP

1. Open PatchMix DSP application.
2. Insert an ASIO Input mixer strip into PatchMix DSP. (Alternately, you can select "New Session", select "E-Wire Example" and skip to step 6.)
3. Mute the strip or turn the Fader all the way down.
4. Insert an ASIO Send plug-in into one of the inserts on your ASIO strip.
5. Name your ASIO strip as an E-Wire strip.
6. Insert the desired PatchMix DSP effects into slots above the ASIO Send.
7. Save the Session.

Setup Cubase

8. Launch Cubase.
9. Instantiate E-Wire in an Insert or Aux Send location within Cubase.
10. Edit the E-Wire plug-in and activate the plug-in by pressing the blue button.
11. Set the ASIO Send and Return on the E-Wire plug-in to match the strip you set up for E-Wire.
12. Done.

E-Delay Compensation

An E-Delay Compensator must be inserted into any other audio tracks that are not using E-Wire in order to keep them time-aligned.

13. Simply insert an E-Delay Compensator plug-in into the same insert location you used for E-Wire on any other audio tracks. That's it.

E-Delay Compensator

As audio is transferred back and forth between the VST host application and the E-MU sound hardware, a delay in the audio stream is incurred. Normally this delay is compensated for automatically by the host application, but not all VST host applications support this automatic compensation.

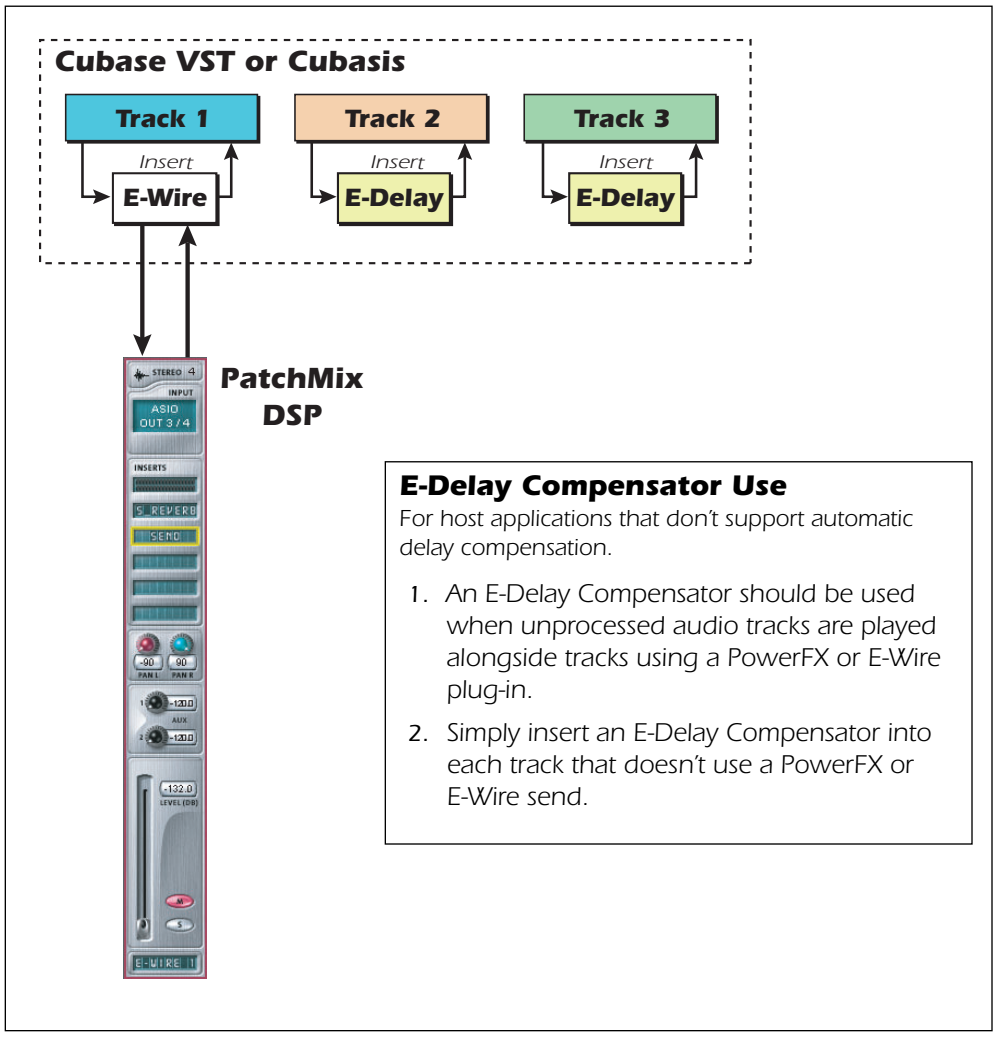
A host will support PowerFX and E-Wire's plug-in delay compensation if it supports the SetInitialDelay feature of the VST 2.0 specification.

Currently automatic delay compensation is supported by the Steinberg 2.0 family (Nuendo 2.x, Cubase SX 2.0, Cubase LE 2.0,), Magix Samplitude 7.x, and Sonar (using the Cakewalk VST adapter 4.4.1), but not by Steinberg Cubase VST 5.1 and Cubasis.

The E-Delay Compensator utility plug-in is used to manually compensate for the transfer delay for hosts that DO NOT support plug-in delay compensation.

The E-Delay Compensator plug-in is used to delay the "dry" tracks (tracks without a PowerFX or E-Wire as an insert effect) or auxiliary (send) channels. For each dry track or send, add an E-Delay Compensator plug-in to re-align the track. The E-Delay Compensator is automatic and requires no user interaction to operate.

For example, consider a Cubasis session with two audio tracks. If PowerFX or E-Wire is applied as an insert effect to the first audio track, but not to the second, the first track will be delayed in relation to the second track. The E-Delay Compensator should be added as an insert effect on the second track in order to provide delay compensation.

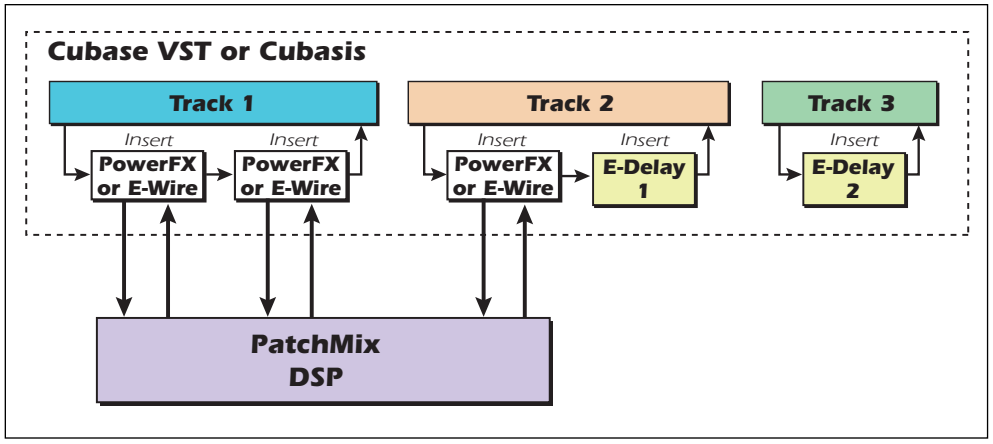

Cubase VST or Cubasis
Track 1
Track 2
Track 3
Insert
E-Wire
Insert
E-Delay
Insert
E-Delay
PatchMix
DSP
E-Delay Compensator Use
For host applications that don't support automatic delay compensation.
1. An E-Delay Compensator should be used when unprocessed audio tracks are played alongside tracks using a PowerFX or E-Wire plug-in.
2. Simply insert an E-Delay Compensator into each track that doesn't use a PowerFX or E-Wire send.

E-Delay Units Parameter

The Units value in the E-Delay dialog box should be set for the number of times you send ASIO down to the PatchMix DSP mixer and back in a single track. A single PowerFX insert chain with any number of effects only requires one delay unit because there was only one trip to the hardware and back. If you use two Cubasis inserts in series on a track both using PowerFX or E-Wire, you would set the number parameter to 2 on all other audio tracks. Each trip down to PatchMix DSP and back to Cubasis equals one unit.

In practical use, however, you'll probably never need to use more than one E-Wire VST on a single track since PowerFX effects can be placed in series. We have included this feature "just in case" you need it.

Here's one more example of how to use the E-Delay Compensator with different numbers of PowerFX/E-Wire sends on each track. The delay compensation on each track must equal the track with the maximum number of PowerFX/E-Wire sends. See the diagram below.



Since track 1 uses two PowerFX/E-Wire inserts, the delay of all the other tracks must equal two. Track 2 has one PowerFX/E-Wire insert and so adding one unit of E-Delay keeps it time aligned. Track 3 doesn't use a PowerFX/E-Wire insert and so needs two E-Delay Units to remain in alignment.

Grouping Tracks

When several tracks require E-Delay Compensation, you can send the output of each track to a group or bus and use a single E-Delay Compensator on the output of the group or bus.

- E-MU Digital Audio System and PatchMix DSP must be installed.
- E-Wire is compatible with Cubase SX/SL/LE, Cubase VST, Wavelab, and Cakewalk Sonar (via DirectX-VST adapter) among others.

6 - Appendix

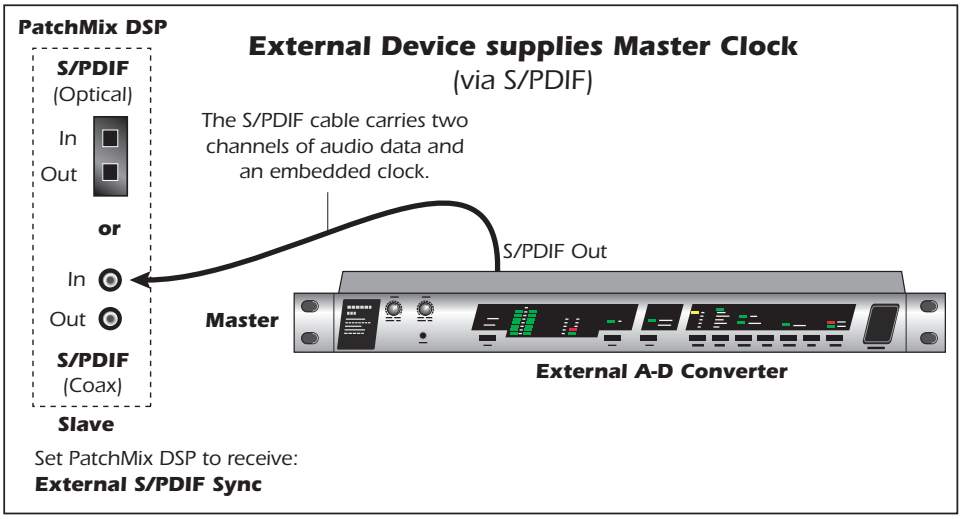
Getting in Sync

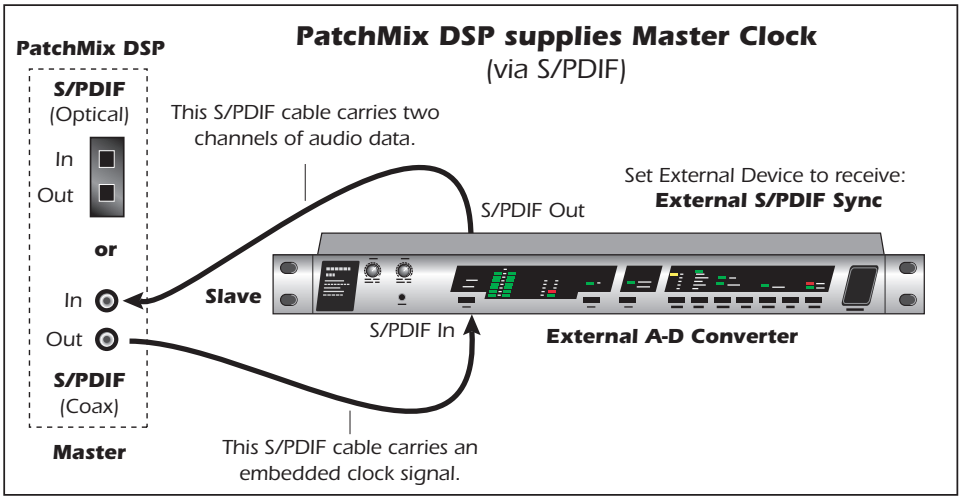
Whenever you connect external digital audio devices together, you need to be aware of how they are synchronized to each other. Simply connecting digital out to digital in doesn't guarantee that two digital devices are synced, even if audio is being passed. Unless you have set one to be the Master and the other a Slave, they are probably NOT synchronized and the quality of your audio will suffer.

S/PDIF is probably the most common digital audio format. S/PDIF carries an embedded word clock which can be used to synchronize the digital equipment. You must enable "External Clock" on the slave device to have clock sync!

The diagrams below show two ways to synchronize an external A/D converter to the E-MU Digital Audio System using the S/PDIF connection.

In the first example, the external A/D converter is the master clock for the system. Only one S/PDIF cable is needed (optical or coaxial) as long as PatchMix is set to receive its word clock signal from the external device. The external A/D is the Master and the E-MU DAS is the Slave.



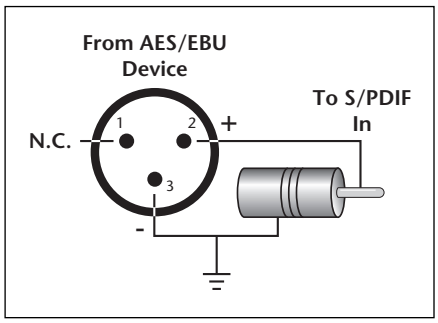


In the second example a second S/PDIF cable is used to supply "embedded word clock". The external device MUST be set to receive external clock via S/PDIF or the units will not be synchronized. The E-MU Digital Audio System is the Master and the external A/D is the Slave.

Useful Information

AES/EBU to S/PDIF Cable Adapter

This simple adapter cable allows you to receive AES/EBU digital audio via the S/PDIF input on the E-MU 0404 PCIe card. This cable may also work to connect S/PDIF out from the 0404 digital breakout cable to the AES/EBU input of other digital equipment.



Digital Cables

Don't cheap out! Use high quality optical fiber and low-capacitance electrical cables when transferring digital I/O to avoid data corruption. It's also a good idea to keep digital cabling as short as possible (1.5 meters for plastic light pipes; 5 meters for high quality glass fiber light pipes).

Grounding

In order to obtain best results and lowest noise levels, make sure that your computer and any external audio devices are grounded to the same reference. This usually means that you should be using grounded AC cables on both systems and make sure that both systems are connected to the same grounded outlet. Failure to observe this common practice can result in a ground loop. 60 cycle hum in the audio signal is almost always caused by a ground loop.

Appearance Settings in Windows

Adjusting the "Performance Options" in Windows will improve the screen appearance when moving the mixer around on the screen.

▶ To Improve the Appearance Settings:

1. Open the Windows Control Panel. *(Start, Settings, Control Panel)*.
2. Select System. Select the Advanced Settings tab.
3. Select Settings in the Performance section.
4. Under Visual Effects, select Adjust for Best Performance. Click OK.

Technical Specifications

GENERAL	
Sample Rates	44.1 kHz. 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz and 192kHz derived from internal crystals. *(No sample rate conversion is performed.)* Externally supplied clock from S/PDIF.
Bit Depth	16-bit or 24-bit *(depending on the setting of your recording or audio application)*
Hardware DSP	100MIPs custom audio DSP. DSP - 32-bit integer math with a 67-bit accumulator Zero-latency direct hardware monitoring with effects
PCIe Specification	▪ PCIe base specification 1.1 compliant ▪ Form Factor: Universal Keyed, PCIe x1 card ▪ 3.3V I/O ▪ PCIe Bus-mastering DMA subsystem reduces CPU use
Converters & OpAmps	ADC - PCM1804 (TI/Burr-Brown) DAC - AK4395 (AKM) OpAmp - NJM2068M (JRC)
ANALOG LINE INPUTS	
Type	Unbalanced, low-noise input circuitry
Level	Consumer: -10 dBV nominal, 6.4 dBV maximum
Frequency Response	20 Hz - 20 kHz: +0.20/-0.10 dB
THD + N	-100 dB (.001%) 1kHz at -1 dBFS
SNR	111 dB (A-weighted 22kHz BW)
Dynamic Range	111 dB (1kHz, A-weighted, 22kHz BW)
Channel Crosstalk	< -120 dB, (1 kHz signal at -1 dBFS)
Input Impedance	3.3K ohm
ANALOG LINE OUTPUTS	
Type	Unbalanced, low-noise circuitry
Level	Consumer: -10dBV nominal, 6.4dBV maximum
Frequency Response	+0.05/-0.10 dB, (20 Hz - 20 kHz)
THD + N	-100 dB (.001%) 1kHz signal at -1dBFS
SNR	116 dB (A-weighted, 22 kHz BW)
Dynamic Range	116 dB (1 kHz, A-weighted, 22 kHz BW)
Stereo Crosstalk	< -109 dB, (1 kHz signal at -1 dBFS)
Output Impedance	560 ohms

DIGITAL I/O	
S/PDIF	▪ 2 in/2 out coaxial (transformer coupled output) ▪ 2 in/2 out optical ▪ AES/EBU or S/PDIF (switchable under software control)
MIDI	1 in, 1 out (16 MIDI channels)
SYNCHRONIZATION	
Internal Crystal Sync:	44.1kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz, 192 kHz
	▪ S/PDIF (optical or coaxial)
RMS JITTER @ 44.1K (Measured via Audio Precision 2)	SRSync SourceRMS jitter in picoseconds 44.1 Internal Crystal 596ps 44.1 Optical Input 795ps

Dimensions & Weight

0404 PCIe Card	
Weight:	0.25lb / 0.10kg
Dimensions:	**L**: 156mm **H**: 107mm

Internet References

The internet contains vast resources for the computer musician. A few useful sites are listed here, but there are plenty more. Check it out.

Software Updates, Tips & Tutorials	http://www.emu.com
Setting up a PC for Digital Audio	http://www.musicxp.net
MIDI Basics	Search for "MIDI Basics" (many sites)
MIDI & Audio Recording	http://www.midiworld.com
MIDI & Audio Recording	http://www.synthzone.com
ASIO, Cubase & Digital Audio	http://www.steinberg.net
ASIO, Cubase & Digital Audio	http://www.steinbergusers.com/cubasele/le_support.php
Cubase Users Group	http://www.groups.yahoo.com/group/cubase/messages

Forums

Unofficial E-MU Forum	http://www.productionforums.com/emu/
KVR Forum	http://www.kvr-vst.com/forum/search.php
Driver Heaven Forum	http://www.driverheaven.net/search.php?s
MIDI Addict Forum	http://forum.midiaddict.com/search.php
Home Recording Forum	http://homerecording.com/bbs/search.php?s=d866b60193933eb726660e7bd90dfb27
Sound-On-Sound Forum	http://soundonsound.com
Studio-Central Cafe Forum	http://studio-central.com/phpbb/search.php
Sound Card Benchmarking	http://audio.rightmark.org

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